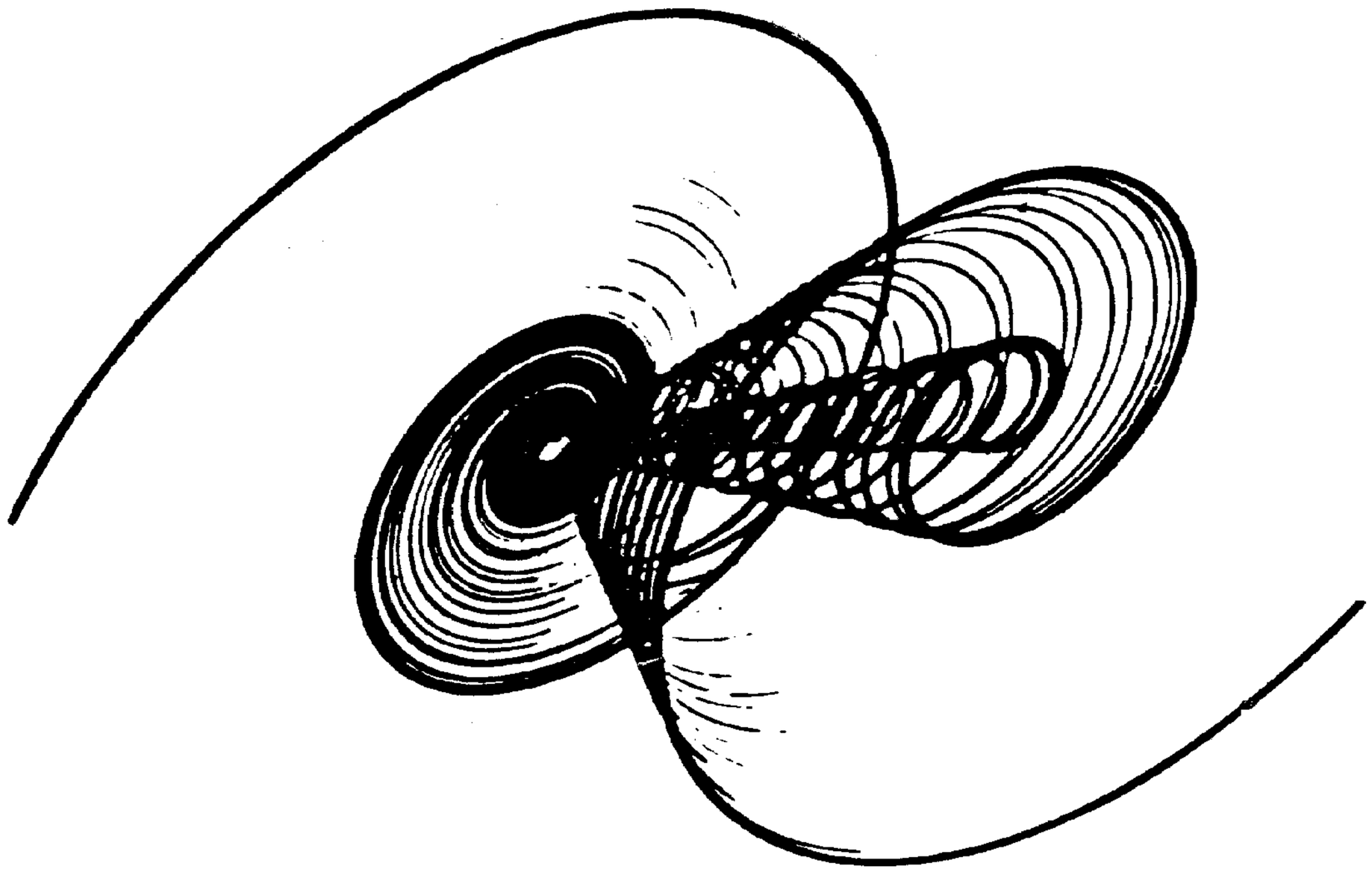


OWNER'S MANUAL

THE ARP SYNTHESIZER

SERIES 2600

DO'S AND DON'TS

1. DO STUDY THE INSTRUCTION MANUAL CAREFULLY AND OFTEN.

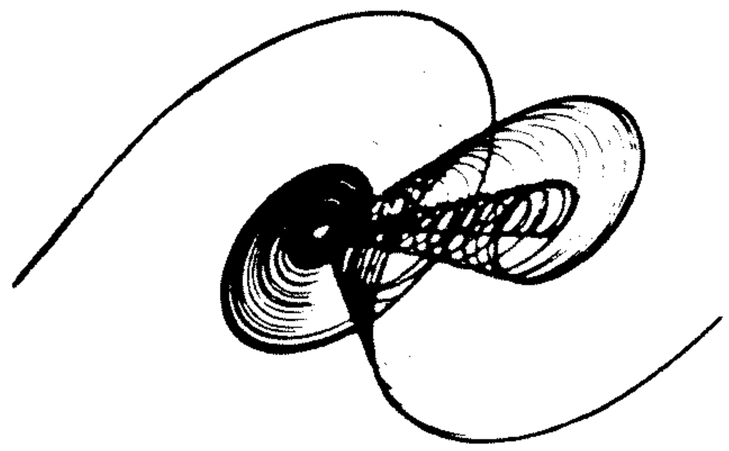
If you are extremely impatient to begin making music with your 2600, you can turn directly to section 5 of the Manual. Use the two panel charts from the Manual as guides. But only a careful study of the Instruction Manual will show you the infinite possibilities for sound production that are contained in the 2600.

2. DO NOT ATTEMPT TO ADJUST THE RECESSED CALIBRATION AND ALIGNMENT POINTS ON THE FRONT PANEL.

until you are thoroughly familiar with your unit and with section 6 of the Manual. If you have any questions about calibration, or any doubts about your abillty to perform the alignment procedure without assistance, contact your Factory-Authorized Service Center.

3. DO NOT, UNDER ANY CIRCUMSTANCES, OPEN UP THE SYNTHESIZER OR IN ANY WAY MEDDLE WITH ITS INSIDES.

This will immediately void your warranty.



OWNER'S MANUAL

THE ARP ELECTRONIC MUSIC
SYNTHESIZER
SERIES 2600

tonus, inc. 45 Kenneth Street, Newton Highlands, Massachusetts. 02161
(617) 969-0810. Cable: ARPSYN

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Printed in USA, February 1971.

tonus inc.

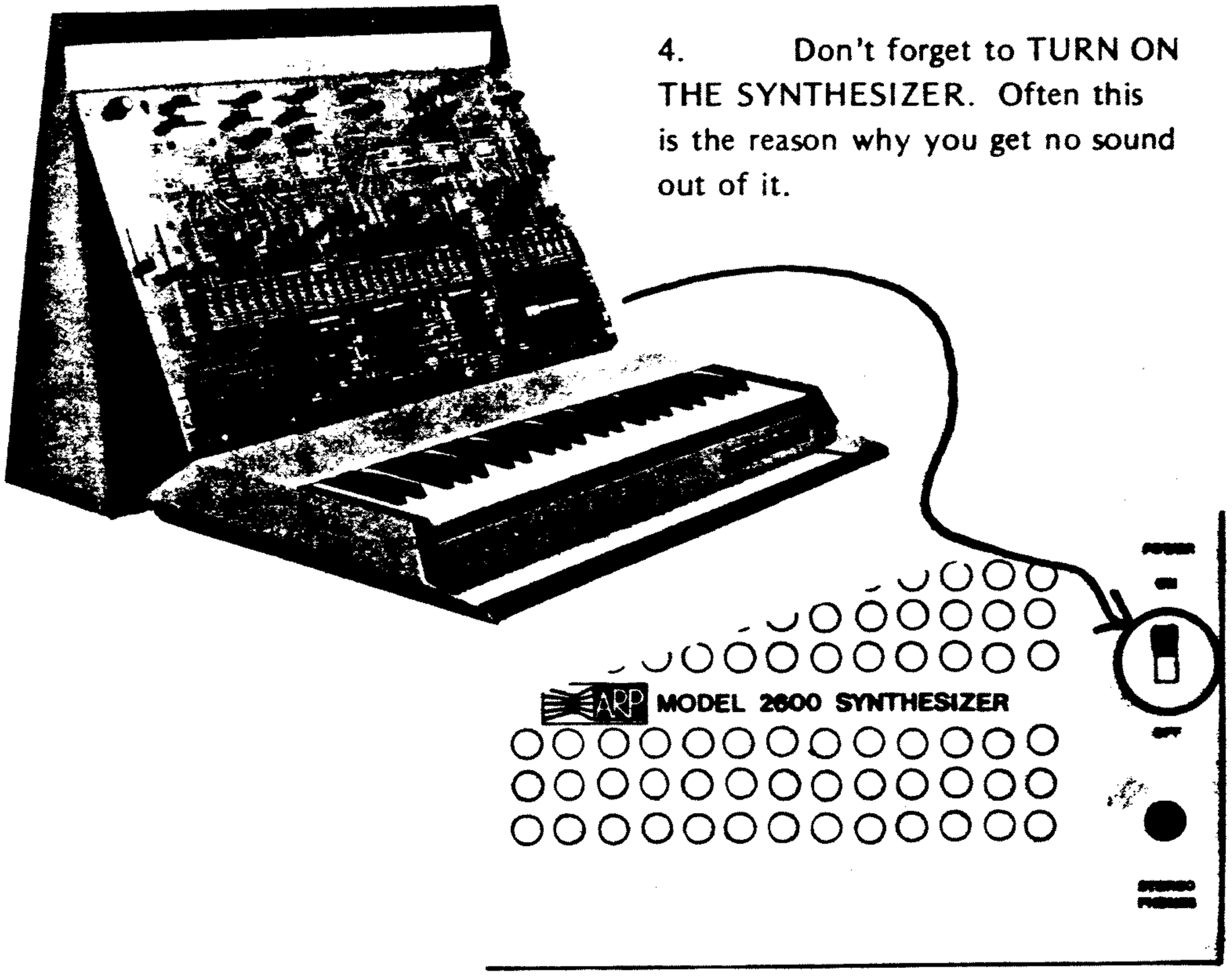
INSTRUCTIONS FOR UNPACKING AND SETTING UP THE ARP 2600.

1. Remove the main unit from its packing case and set it on a firm support, preferably against a wall or other backing.

2. Remove the 3604 keyboard unit from its case and set it in front of the 2600. Two cable connectors are taped to the back of the keyboard. Remove the tape and plug them into the sockets on either side of the 2600 near the bottom.

3. A grounded power cable is supplied with the 2600. Plug it into the three-pin plug on the back of the main unit. The 2600 is designed to operate from 110VAC sources. Do not attempt to use it from a DC source.

4. Don't forget to TURN ON THE SYNTHESIZER. Often this is the reason why you get no sound out of it.



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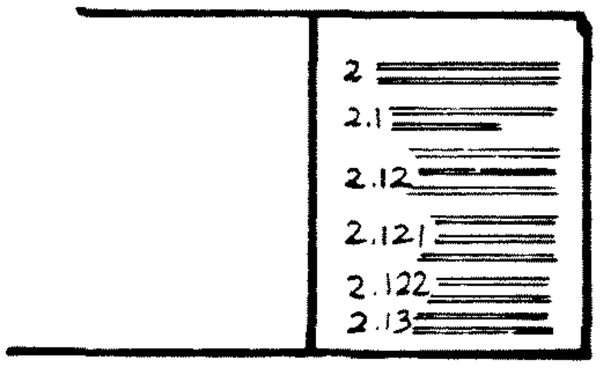
1.0 THIS MANUAL is designed to serve two purposes.
1.01 THE FIRST is to serve as an instruction book for the ARP Model 2600 Electronic Audio Synthesizer.
1.02 THE SECOND is to serve as a general introduction to the terms, concepts, and practices involved in electronic sound synthesis. We have designed it to serve this purpose because, first of all, there is at present no such general introduction available, and second because the ARP 2600 is specifically designed for those who have no prior experience either in using synthesizers or in thinking about sound in the peculiar way that working with a synthesizer leads one to think.



1.1 THE PARAGRAPHS in this manual are numbered decimally to correspond with the structure of topics. The most important general headings are numbered with a single digit; subtopics under each heading will follow the digit with a decimal point and another digit, and so on.

1.2 THE GENERAL HEADINGS and their numbers will be as follows.

1 – THE MANUAL.
2 – ELECTRONIC SOUND SYNTHESIS AND SYNTHESIZERS.
3 – THE ARP 2600. A general description.
4 – THE ARP 2600 FUNCTIONS. Individually.
5 – USING THE ARP 2600. Or: how to put it all together.
6 – CALIBRATION, MAINTENANCE, ADJUSTMENT. Without tears.
7 – APPENDIX

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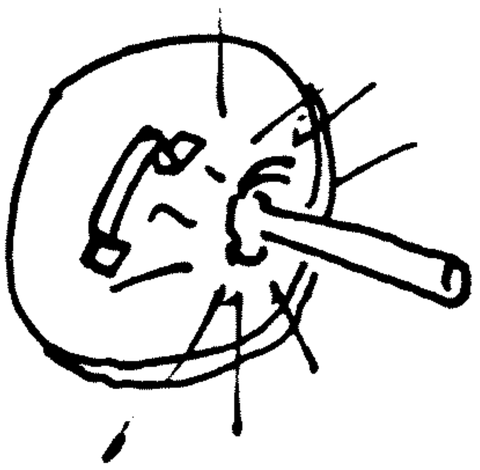
2. TWO BASIC IDEAS ARE INVOLVED in all electronic music synthesizers.

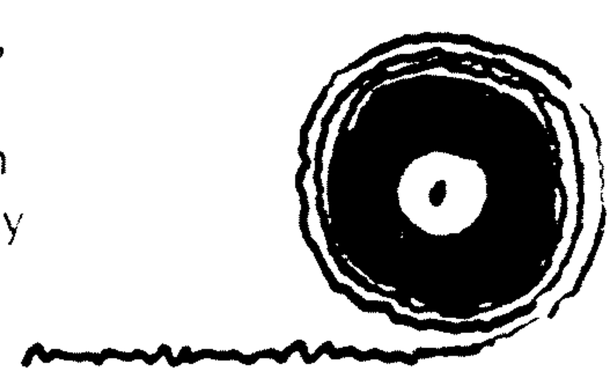
2.1 THE FIRST is that ACOUSTICAL WAVEFORMS CAN BE GENERATED AND MODIFIED PURELY BY ELECTRONIC MEANS.

2.11 Banging on a garbage can lid generates a horrible racket by mechanical means. The racket is a very complicated sound; but no matter how complicated it gets, it can be reproduced by a single long and complicated wiggle in a phonograph record. So can a symphony. From a certain point of view all the sounds you've ever heard, ever will hear, and ever could imagine, must be reducible to one (or at most two—one for each ear) complicated wiggle of your eardrums.



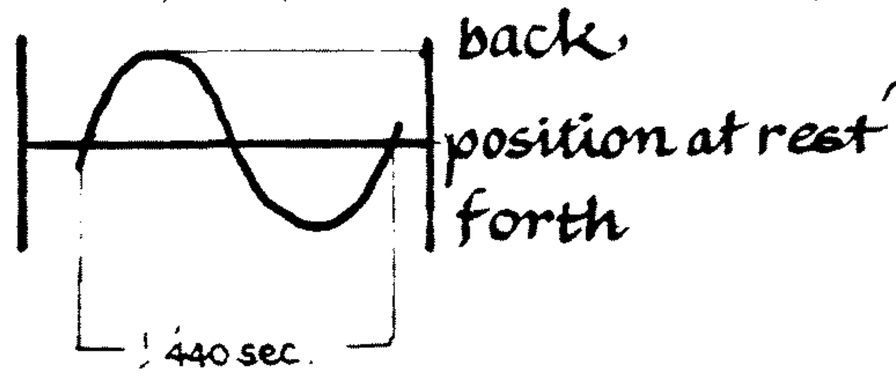
2.111 Edison's phonograph recorded by mechanical means the vibrations he fed into it. The cylinder—and later the disc—stored them as mechanical wiggles in a groove; on playback, running this groove past a needle made the needle wiggle, the needle made a diaphragm vibrate, and the diaphragm passed it on to the air in the form of pressure vibrations, and the air transmitted the sound waves to somebody's ears.

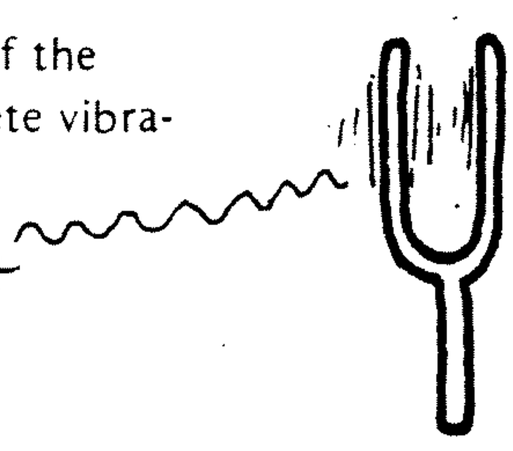
2.112 Nobody has done it that way for quite a while now. Today, a microphone turns sound waves into electrical signals. In other words, rapid and periodic fluctuations in air pressure are transformed into rapid and periodic fluctuations in some electrical phenomenon—usually a voltage. In this form the wave can be electrically amplified, equalized, filtered, and subjected to various other indignities before being sent to a recording lathe that turns it back again into a mechanical waveform in a groove. Likewise, when the record is played, the stylus wiggles the way it did in Edison's player, but its vibrations are immediately turned into electrical signals. They don't become mechanical again until they reach the loudspeaker.

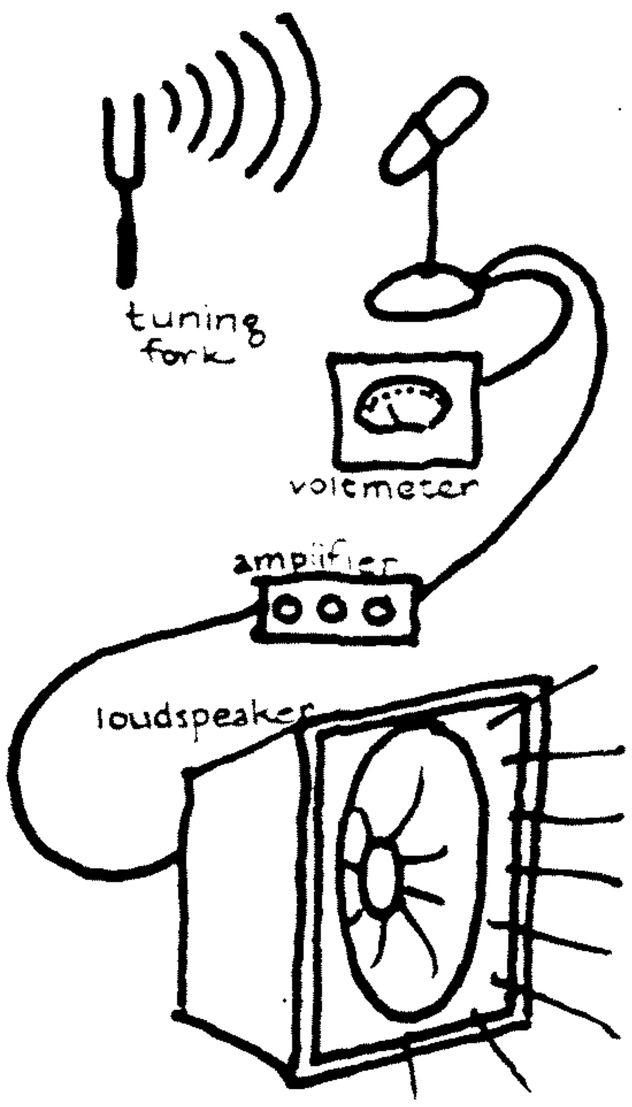


2.12 IN ALL THIS, THE ONLY THING THAT DOESN'T CHANGE IS THE SHAPE OF THE WAVE, i.e. THE WAVEFORM.

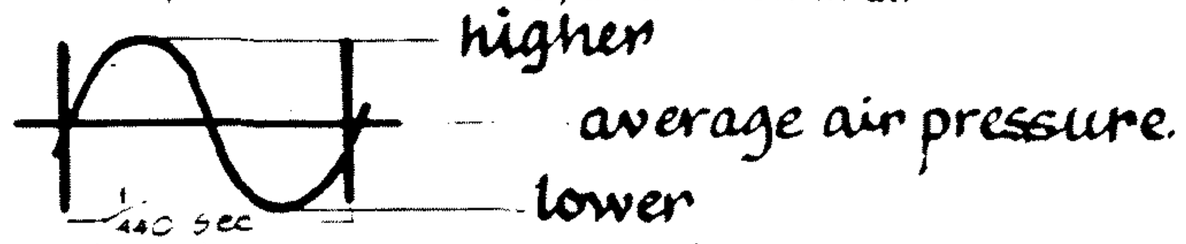
2.121 Suppose you bang an A-440 tuning fork on the corner of the table. In 1/440th of a second, then, the fork makes one complete vibration back and forth:



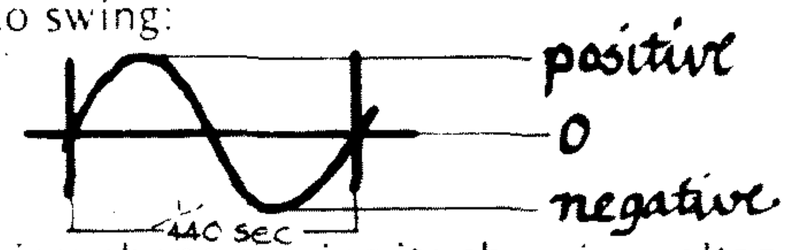




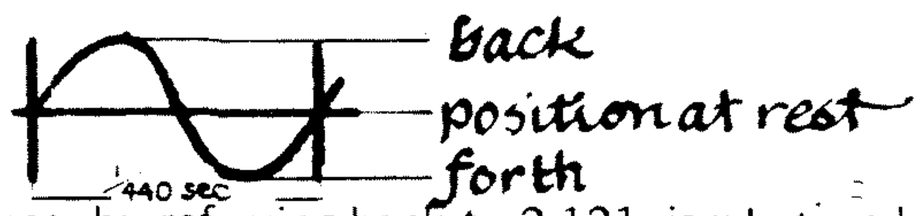
2.122 If you had an exceedingly precise barometer around, it would register during that same 1/440th of a second, a variation in air pressure:



2.123 And a nearby microphone connected to a voltmeter would cause the pointer to swing:



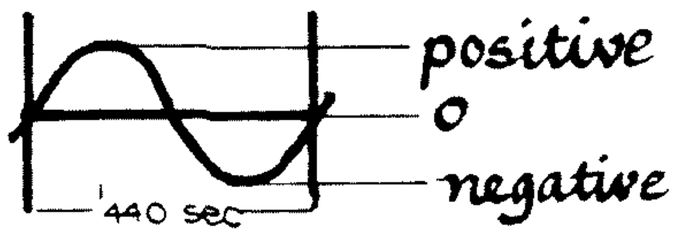
2.124 The same microphone passing its changing voltages on to an amplifier, and the amplifier feeding its output to a loudspeaker, would make the loudspeaker cone move back and forth:



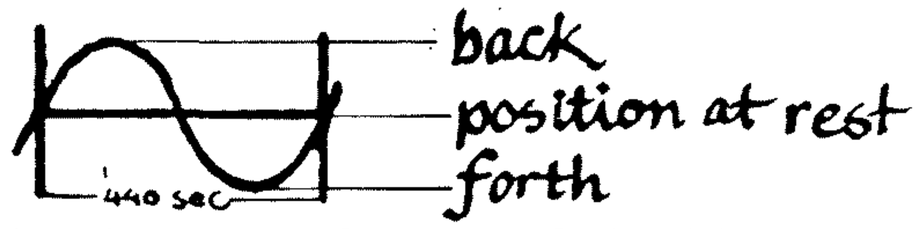
. . . which you will see, by referring back to 2.121, is what we began with.

2.1241 We have used an extremely simple waveform for these examples; but remember that the process of translating waveforms we have just outlined holds for any waveform whatsoever. In our graphs, the only change from one example to the next has been in the meaning of the vertical dimension of the graph; the horizontal dimension always represents some period of time, while the vertical one changes from representing physical position (the "back and forth" of the tuning fork or the loudspeaker cone) to a physical quantity (air pressure) to an electrical quantity (positive or negative voltage). But the waveform is unchanged.

2.125 Take another look at what we said in 2.123 and ask yourself what the voltmeter would do if instead of being connected to the microphone it were connected to some kind of electrical circuit that could generate smoothly changing voltages at the rate of 440 times each second. The pointer would swing positive and then negative like this:



2.126 And the same generating circuit hooked up to the amplifier hooked up to the speaker would cause the same back-and-forth movement as in 2.124:

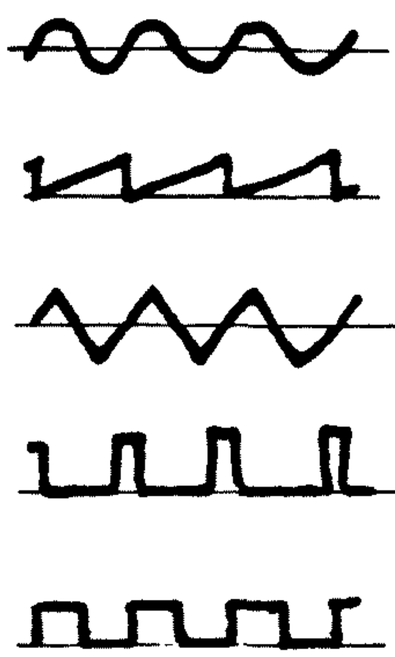


. . . which amounts to getting the sound of a tuning fork without having a tuning fork. Or, by the same principle, the sound of somebody banging on a garbage can lid without either a garbage can lid or somebody to bang on it. Or any sound at all: if you can generate the right waveform electronically, then you can generate the sound.

2.127 The generators we talked about in 2.125 are not imaginary; they exist, and they are called OSCILLATORS, and every synthesizer has at least a few of them; a really big synthesizer might have several dozen. So do electronic organs.

2.13 IF THE SHAPE OF THE WAVEFORM IS MODIFIED, SO IS THE SOUND, and vice versa. Take this for the time being as an ironclad law; there are exceptions but they don't matter just now.

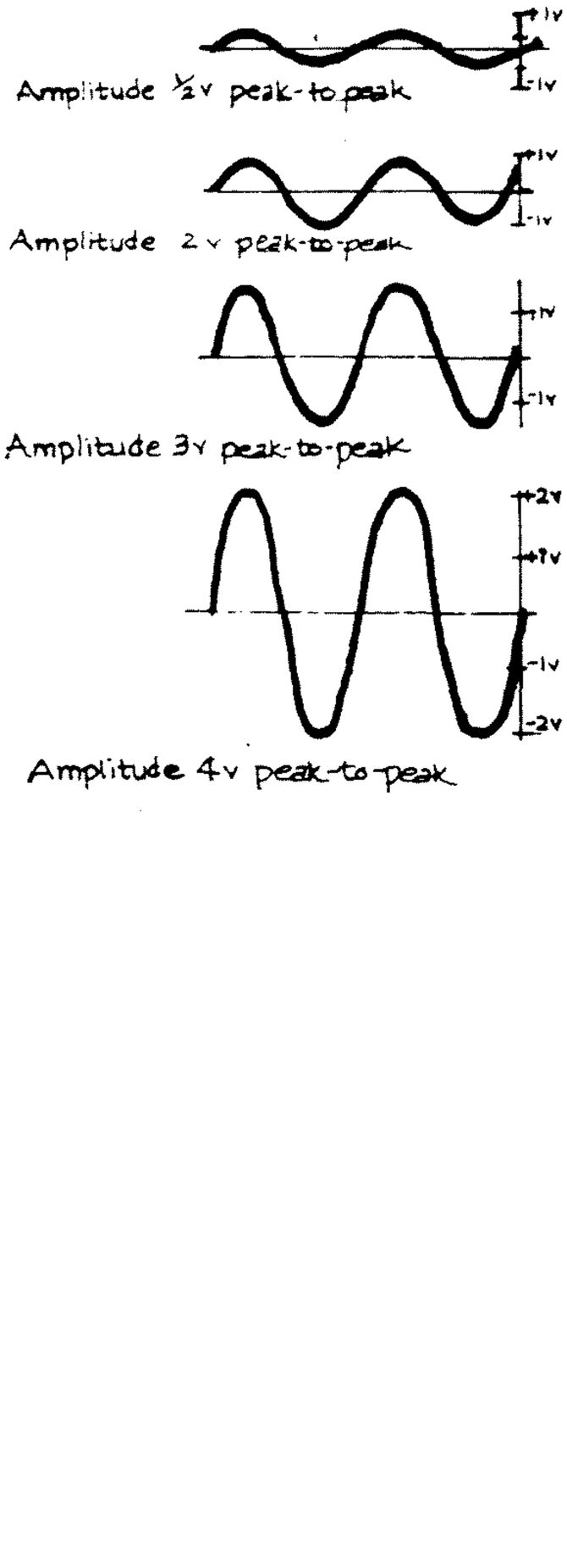
2.131 Wrapping an old army blanket around the garbage can lid muffles the sound. Carrying it, wrapped or unwrapped, into a tiled bathroom while you're banging on it changes the sound again; and so on. Anything you can do mechanically to modify the sound is going to show up as a modification in the pattern of air pressure variations and therefore in the pattern of the changing voltages produced by the microphone. Turn the can lid into a shoebox and kick it across the room—a different waveform. Turn it into a kettledrum—another waveform. Spread rubber cement over the drumhead—another waveform. Fill it with water—for every pint, another waveform; not by much, maybe, but still different.



2.132 By the same principle, any modification in an electrical waveform will modify the sound it makes through an amplifier and loudspeaker. The tone controls on a phonograph change the electrical waveforms that pass through them; changing stops on an electronic organ or drawbars on a Hammond changes the electrical waveform produced; and EVERY SINGLE CONTROL ON A SYNTHESIZER has some effect on an electrical waveform.

2.14 THE TWO SIMPLEST RELATIONSHIPS BETWEEN WAVE-FORM-CHANGES AND SOUND-CHANGES are these:

2.141 Increasing the AMPLITUDE of an audio waveform increases the VOLUME of the sound it makes. (But it might make no sound at all—the amplifier might be turned off or the loudspeaker disconnected. The point of this joke is to remind you that electrical vibrations and sound waves are not, after all, the same kind of activity; there is no sound inside electronic organs, or synthesizers either. Sound doesn't enter into the picture at all until the rapidly fluctuating voltages which the organ or synthesizer generates are amplified and fed to a loud-speaker.) (YOU are interested only in the sounds—but the synthesizer doesn't know that.)

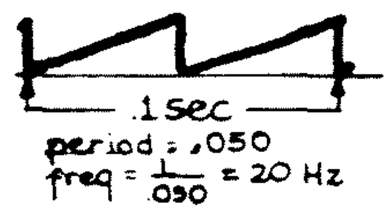


2.1411 The AMPLITUDE of a waveform is the amount of maximum deviation from its "center". In a loudspeaker this "center" is the position of the loudspeaker cone at rest; in an electrical circuit it might be a condition of zero voltage. The amplitude of a loudspeaker cone's motion would be measured in inches or more likely in fractions of an inch back and forth. The amplitude of a fluctuating or alternating voltage would be measured in volts positive and negative. Thus a voltage waveform that reached a peak of +1 V and then of −1 V would have an amplitude of 2 V "peak-to-peak".

2.1412 You'll save yourself a lot of confusion by thinking of AMPLITUDE only in connection with waveforms and of VOLUME only in connection with sounds. For example: because there are no sounds inside a synthesizer (see 2.141), there cannot logically be any volume controls; but there are a great many amplitude controls. The point of our putting so much emphasis on this distinction will become clearer as you read on.

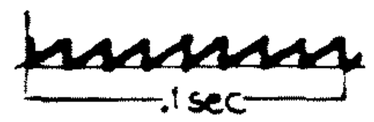
2.142 Increasing the FREQUENCY of an audio waveform increases the PITCH of the sound it makes. But it is important here, just as with AMPLITUDE and VOLUME, to keep the two notions distinct. FREQUENCY is a characteristic of PHYSICAL VIBRATIONS (whether

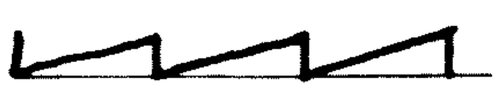
mechanical, electrical, or otherwise), but PITCH is a peculiar characteristic of THE WAY HUMAN BEINGS PERCEIVE physical vibrations between approximately 20Hz and 20KHz. And of course the whole point of human perception anyway is to register and take note of changes in one's surroundings. So a change in an audio frequency registers as a change in pitch, but frequency and pitch are not the same.

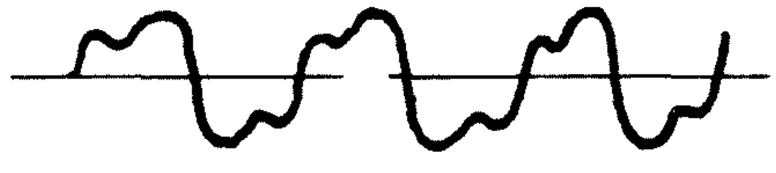


2.1421 For example: every pitch is produced by some frequency, but many frequencies produce no sense of pitch at all—because they're not audio frequencies. Middle C on a piano corresponds with a frequency of about 261 cycles per second—but a frequency of 2.61 cycles per second corresponds to no pitch at all—it is far below audibility.

2.14211 Here is some useful terminology. Some kinds of vibration have a repeating pattern;

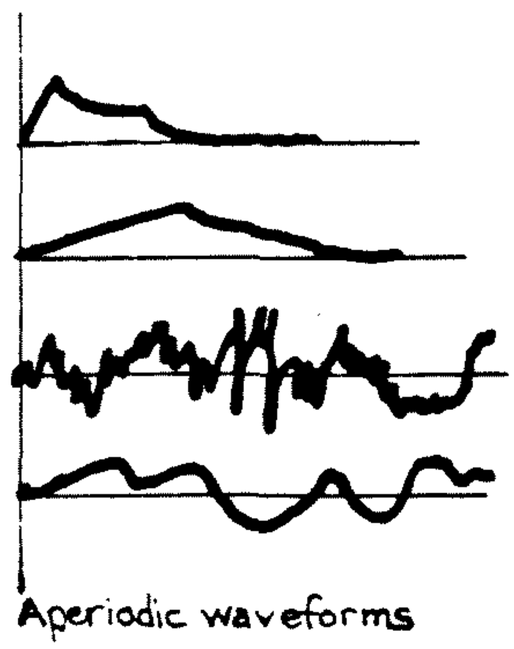






and if they do they are called PERIODIC. One segment of a periodic vibration, from any point in its waveform to the beginning of its repetition, is a CYCLE. The length of one cycle (usually stated in seconds or in fractions of a second) is the PERIOD of the waveform; but the number of cycles occurring in a given length of time is the FREQUENCY of the vibration. There is an international standard unit of frequency: it is the HERTZ (abbreviated as "Hz") and it is defined as ONE CYCLE PER SECOND. As usual, the prefix "Kilo-" means "one thousand" and so a vibration or oscillation of one thousand cycles per second will be spoken of as "One KiloHertz" or "1KHz".

2.14212 NOT ALL WAVEFORMS are PERIODIC. Some waveforms happen only once; and some waveforms are so complicated that your chances of finding any kind of repetition are nil regardless of what frequency you look for. In this case the waveform as a whole will show no particular repeating pattern but only an apparently random motion. For whatever reason, a waveform that does not demonstrate any repeating patterns is called APERIODIC.

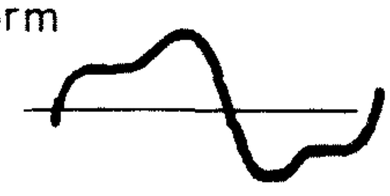


2.14213 We will use these terms throughout the rest of this manual. Frequencies of less than 1Hz will generally be referred to by their period to save you the labor of, say, translating ".05Hz" into "one cycle every 20 seconds".

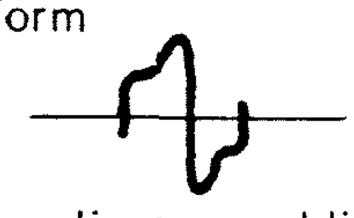
2.15 CHANGES IN THE AMPLITUDE OR FREQUENCY OF A PERIODIC WAVEFORM ARE, INDEED, SO COMMON, AND THEIR RELATIONSHIP TO PERCEIVED SOUND SO CLEAR, THAT THEY ARE NOT COMMONLY THOUGHT OF AS INVOLVING CHANGES IN THE SHAPE OF A WAVEFORM AT ALL.

2.151 From now on we will follow this convention. Changing the frequency or amplitude of a wave will not count as a change in its shape.

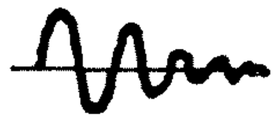
2.1511 There is a more subtle reason than mere convenience for this practice. Graphically, this waveform



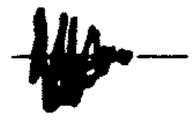
can become this waveform



by a mere change of coordinates; and likewise for turning this waveform



into this one



But no amount of graphical transformation will turn something like this

into this ; no change in mere amplitude or frequency or both or change in units of measurement. (The latter is what we mean by a "change in coordinates".)

"Timbre" is the subjective quality of a tone which enables the listener to distinguish between it and other tones which may have the same pitch or loudness. (e.g. trumpets and clarinets produce sounds of different timbres.)

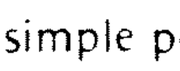
2.152 CHANGES IN THE SHAPE of an audio waveform are generally associated with CHANGES IN THE TONE QUALITY, or TIMBRE,* of the sound produced. (See 2.131). Since human perception has limits, there are possible changes in the shape of a waveform that might not be perceived by even the most practiced ear; but anything that is perceived as a change in timbre must be reflected in the waveform being perceived.

2.1521 Summarizing 2.141, 2.142, and 2.152 gives us these approximate relationships:

subjective changes in perceived...	correspond to physical changes in the...
VOLUME	AMPLITUDE
PITCH	FREQUENCY
TIMBRE or	WAVESHAPE or
TONE COLOR	FREQUENCY-SPECTRUM

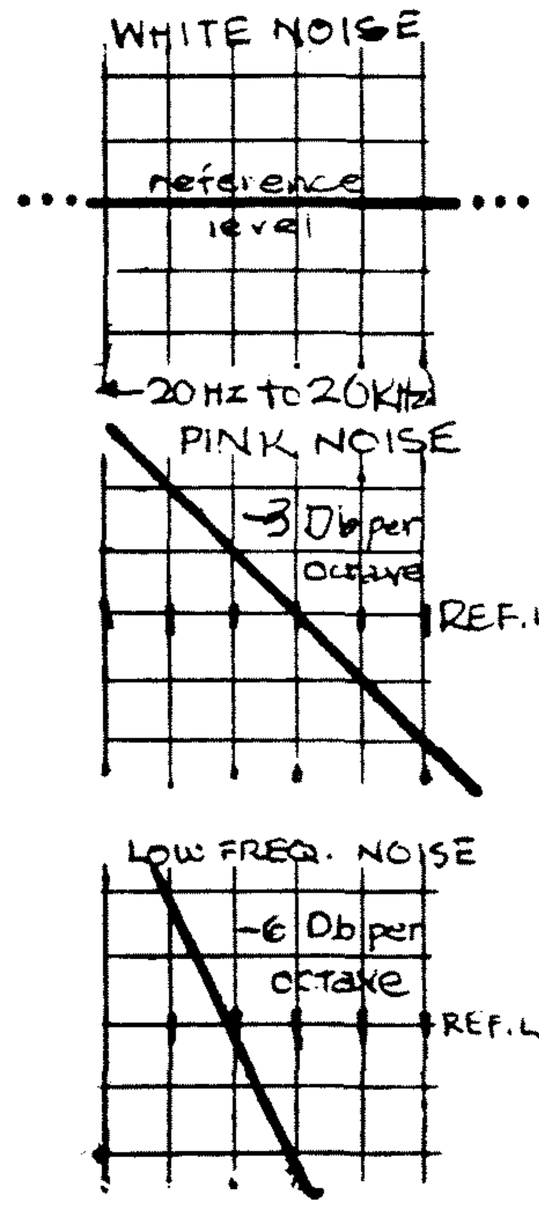
. . . of the perceived vibration.

This is worth remembering because, as we have emphasized elsewhere, synthesizers work with the qualities listed in the right-hand column for the sake of our perception of those listed in the left-hand column; anyone who uses a synthesizer regularly must be able to translate easily from the language of volume, pitch, and timbre, to the language of amplitude, frequency, and waveshapes. This section of the manual is intended to get you started on this sort of translation back and forth.

2.153 The OSCILLATORS in any synthesizer generate one or more of these simple periodic waveforms: SINE  , SAWTOOTH , SQUARE , TRIANGLE , and PULSE .

2.154 A NOISE GENERATOR produces an APERIODIC waveform (see section 2.14212) of completely RANDOM pattern.

2.1541 A random voltage has very interesting properties. You can think of it statistically as a waveform in which your chances of finding any particular frequency are equal to your chances of finding any other frequency. Strictly speaking this sort of a waveform is called WHITE NOISE, by analogy with white light: it contains all frequencies just as white light contains all colors.



2.1542 Human ears tend to give undue prominence (for reasons we needn't go into here) to the higher frequencies in a WHITE NOISE signal, so that it sounds like steam escaping from a radiator. If a white noise signal is slightly FILTERED to produce a noise whose frequency content SOUNDS MORE EQUAL TO HUMAN EARS, it is called PINK NOISE. Pink noise sounds like NIAGARA FALLS.

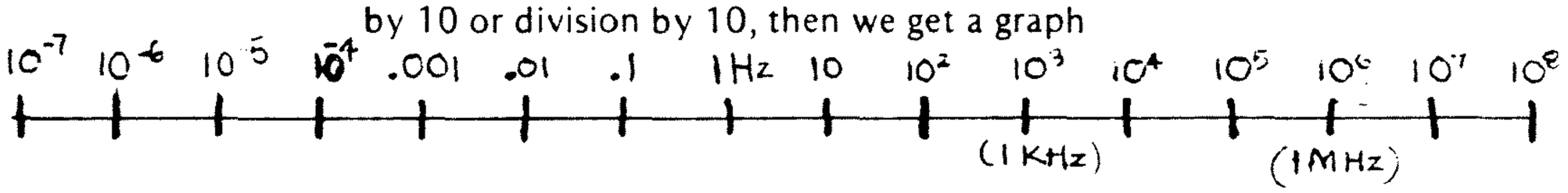
2.1543 From the statistical point of view, what happens when white noise is FILTERED is simply that the odds on finding any particular frequency in the noise waveform become heavily weighted in favor of some certain RANGE OF FREQUENCIES. This range is called the BANDWIDTH of the filtered noise. Thus we can talk about wideband

noise, narrowband noise, noise centered around 2KHz, and so on. (See also 2.182)

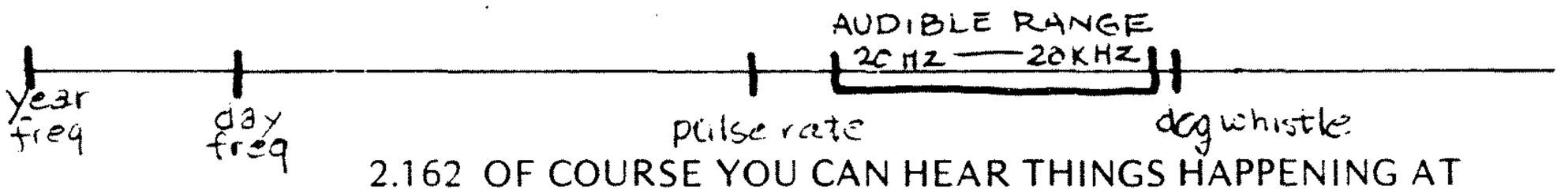
2.155 Each of these waveforms, at audio frequencies, has its own characteristic sound (see 2.13). We will not try to describe here the sound of each waveform; that will come into your own experience in sections 4 and 5.

2.16 We will call any periodic waveform with a basic frequency between 20Hz and 20KHz an AUDIO WAVEFORM. Frequencies between these two extremes are AUDIO FREQUENCIES. Higher frequencies are ULTRA-SONIC and lower frequencies are SUB-SONIC, or simply LOW.

2.161 Our reason for calling some frequencies "audio" is of course simply that only vibrations in this range can produce SOUNDS. If we draw a line and put "1Hz" in the center of it, and let the two ends run off to infinity, and let every half-inch or so represent a multiplication by 10 or division by 10, then we get a graph



in which the range of human hearing is



2.162 OF COURSE YOU CAN HEAR THINGS HAPPENING AT LOWER FREQUENCIES than 20 Hz. BUT YOU HEAR THEM AS SEPARATE AND REPEATING SOUNDS, not as continuing tones or noises.

2.1621 Your heart is beating, let us say, 60 times per minute. (This is rather low for a pulse rate but keeps the numbers simple.) That's the same as saying it beats once every second and so we can say it beats at a FREQUENCY of 1Hz. Let's draw a graph:

Now imagine your pulse gradually increasing to 120 per minute, or 2Hz:

And now to something like 20Hz (that's 1200 times a minute—a lot of adrenalin):

And finally to say 60Hz:

Somewhere just above 20Hz you would lose your sense of individual EVENTS happening very rapidly. Instead you would begin to hear a very low PITCH gradually rising with the rising frequency of your heartbeat. And the opposite would happen if your heart began to slow down again—first a descending PITCH, then the gradually growing sense of no pitch at all but rather of separate and countable EVENTS.

2.163 A SYNTHESIZER CAN GENERATE AND MODIFY both AUDIO-FREQUENCIES (i.e. PITCHES and NOISE) and LOW-FREQUENCIES (i.e. EVENTS).

2.1631 You may think of an event that happens only once as having an infinitely low frequency—like picking a guitar string once for all eternity, or beating a drum just once and then travelling on . . .

2.1632 Playing a note on a piano and repeating it at some regular interval produces a series of EVENTS at a low frequency. The events have a certain PITCH because each event is the occurrence of an AUDIO—FREQUENCY vibration. Here's a graph of everything that happens:

and here's a graph of the LOW-FREQUENCY waveform involved

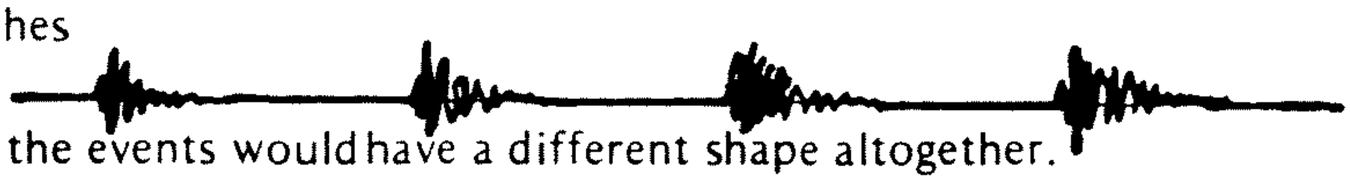
and the audio-frequency that produces the pitch looks, all by itself, like this

2.1633 Note that in any graph of an event or a series of events such as the first one in 2.1631 above, you can derive the SHAPE of the LOW-FREQUENCY WAVEFORM involved by simply connecting the

highest points on the higher-frequency waveform.
2.1634 Thus we come to the very important notion that EVENTS have SHAPES. And the SHAPE of any event is the SHAPE OF THE LOW-FREQUENCY WAVEFORM that can "produce" the event. Thus, for example, playing a staccato tune on an organ would produce a series of pitches

and a series of events of this shape

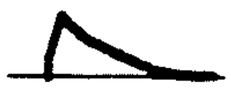
Playing the same tune on a guitar would produce, perhaps, the same pitches



but the events would have a different shape altogether.
2.164 THE SHAPE of an EVENT is called its ENVELOPE or CONTOUR. In the example given above, we could say that the notes played on the organ have a different ENVELOPE from the same notes played on the guitar.
2.1641 Here are some possible envelopes:

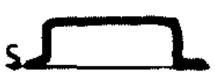
banjo

distant thunder

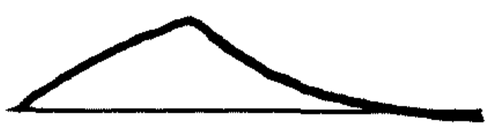
piano note fortissimo 

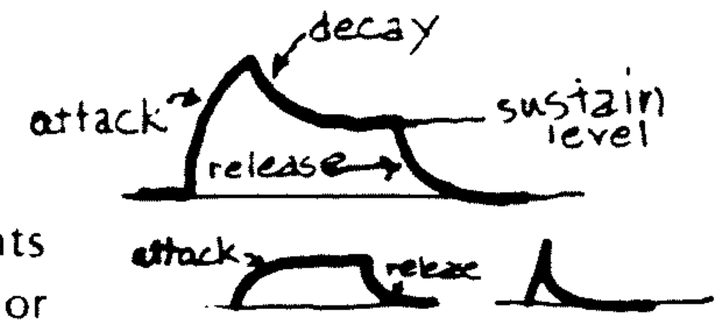
piano note soft and staccato

organ note

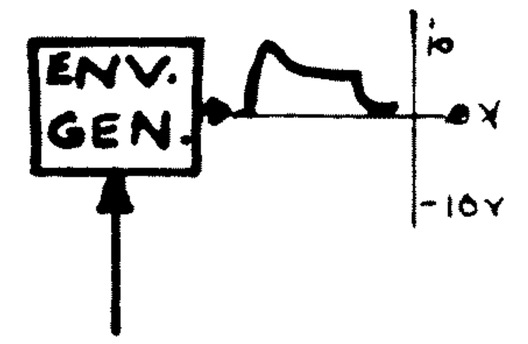
organ note taped and played backwards 

banjo taped and played backwards

Chinese gong 

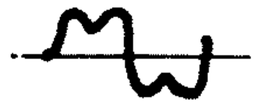


2.17 Any low-frequency waveform may be used to produce events and to give them a shape. But usually a synthesizer will have one or more devices designed specifically to generate low-frequency waveshapes suitable for giving events a shape. These are called ENVELOPE GENERATORS or ENVELOPE TRANSIENT GENERATORS. Their output is APERIODIC (see 2.1422); instead of appearing over and over again spontaneously it appears only when the envelope generator is TRIGGERED.



2.18 OTHER CIRCUITRY in any synthesizer MODIFIES waveforms by MIXING them, FILTERING them, or MODULATING them.

2.181 Two waveforms can be added together by simply adding their values at every instant. This is easier to show than to talk about:

This added to this makes this  . Once they are mixed it's practically impossible, except in simple cases, to untangle them again.

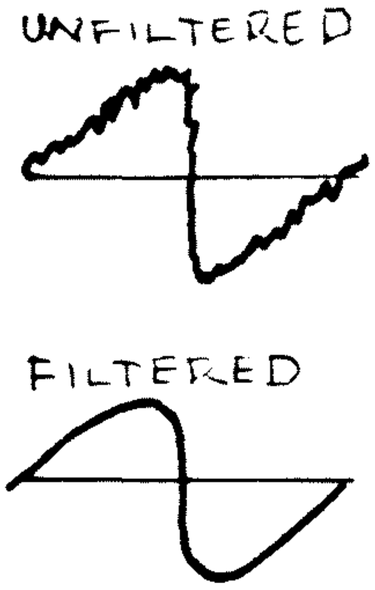
2.1811 IT MAY SEEM ODD that when you have mixed together say the sound of a violin and the sound of a piano (or any two or more instruments) it is all but impossible to separate them again. But remember what is going on: in electrical terms, a particular sound is not represented simply by "a voltage" but by a PATTERN of voltage fluctuations. So when two or more signals are electronically mixed into one, two or more patterns have been added together to form a more complicated pattern. And the problem of separating them is not a simple problem of dividing a voltage into little voltages, but the much more complicated one of separating patterns, i.e. waveforms, out of one complicated pattern.

It is as if you were to copy a page of this manual onto one line by writing each line directly on top of the preceding line:

It is as if you were to copy a page of this manual onto one line by writing each line directly on top of the preceding line: all the letters would be there, but the copy would be illegible. Your ears, from long practice perhaps, generally have no difficulty in recognizing familiar patterns in the 24-hour-a-day waveform the world presents to them; no other instrument can even begin to approach the ease with which, say, you disentangle the voice of a friend in a crowd from all the other voices in the crowd. This is a problem in

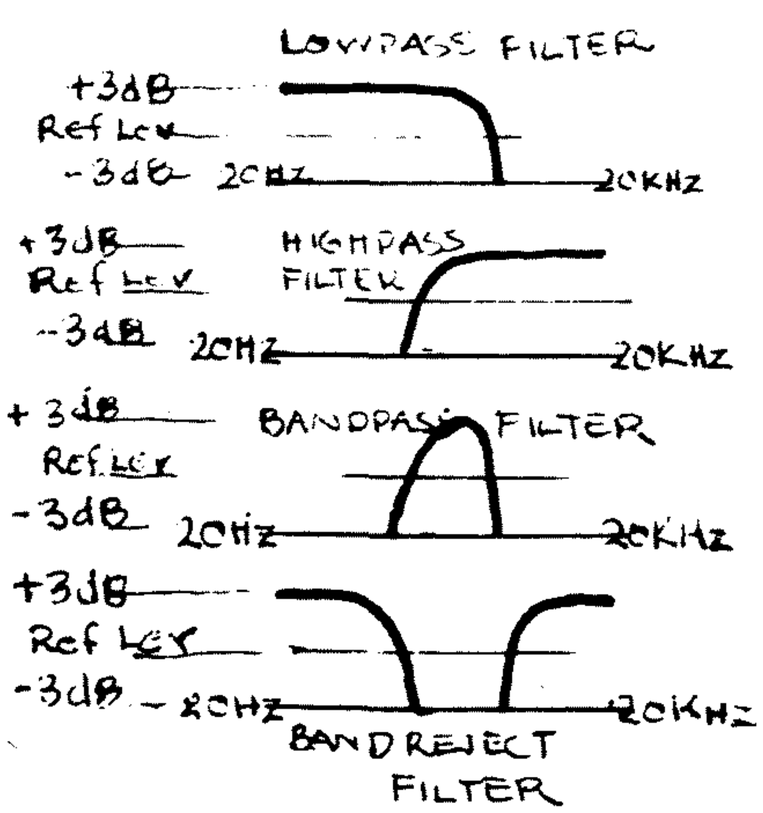
PATTERN RECOGNITION, and has for many years been the subject of intense research by computer-program designers. But the best they have done so far falls considerably short of what your ears and brain do all the time.

2.1812 In very simple cases, frequencies may be removed from a waveform by FILTERING (see 2.182 following), but a filter circuit is a perfect idiot: it knows nothing of what the frequencies it is blocking or boosting mean, and can do nothing to separate two sounds of the same pitch. The scratch filter in a stereo amplifier cuts out some of the noise from a dirty or scratched record; but it also removes any musical sound that falls in the same frequency range as the scratches. On a good pair of speakers this is quite audible as a loss of sparkle and airiness in the recording—a slight muffling of the sound.



2.1813 So one way of making more complicated waveforms out of simple ones is by MIXING simple ones together. This is also sometimes called additive synthesis. The drawbars of a Hammond work by additive synthesis.

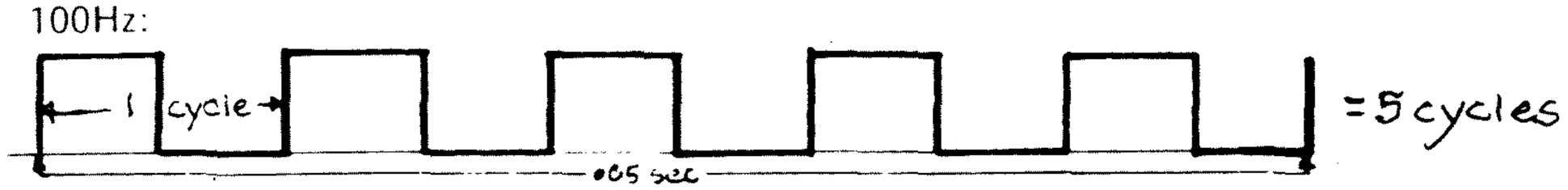
2.182 In a complex waveform representing many frequencies simultaneously, some frequencies may be weakened, strengthened, or removed almost entirely by FILTERING. The treble and bass controls on a stereo amplifier are simple filters that boost or cut high and low audio frequencies respectively. A filter may operate over a broad or narrow range of frequencies; it may pass all signals up a certain frequency, in which case it is called a LOW-PASS filter; it may pass only frequencies above a certain frequency, in which case it is called a HIGH-PASS filter. Or it may pass only a narrow band of frequencies; then it is a BAND-PASS filter. If it does just the opposite, i.e. passes all frequencies except a certain band of frequencies, it is a BAND-REJECT, or NOTCH filter.



2.1821 Simplifying a complex sound by filtering is called subtractive synthesis.

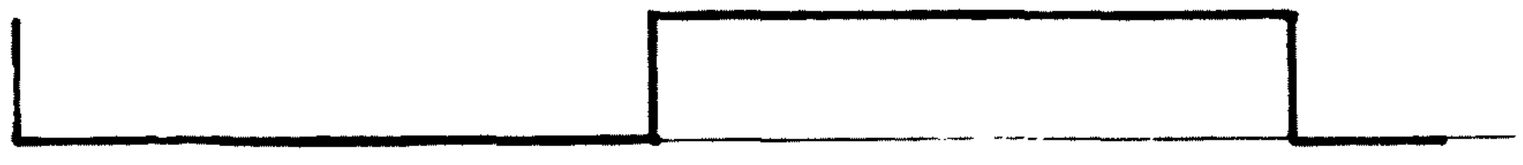
2.183 To MODULATE a waveform is to change it systematically, following the pattern of another waveform. If the change is in frequency, then the result is FREQUENCY MODULATION (FM); if the change is in amplitude, then the result is AMPLITUDE MODULATION (AM). Other kinds of modulation exist and will be discussed later.

2.1831 Here is a simple example of frequency modulation. Suppose we begin with a simple oscillation, say a square wave at a frequency of 100Hz:

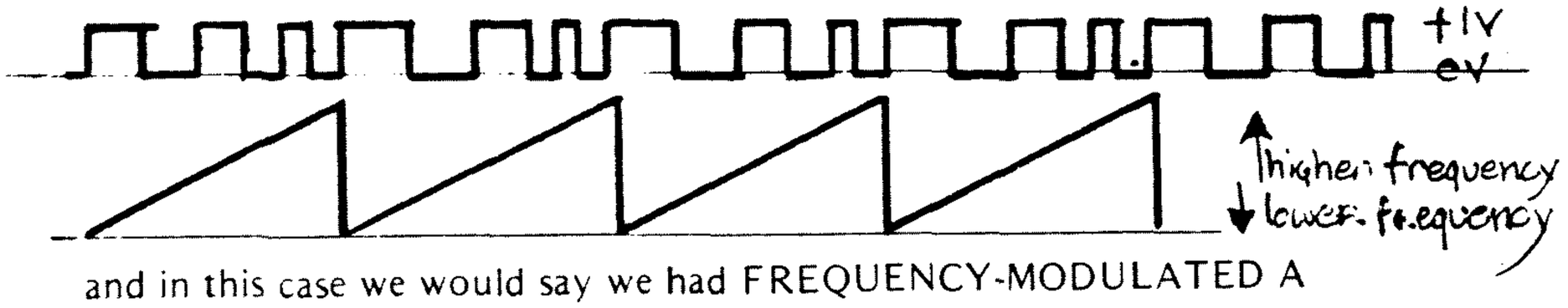


Now, at the rate of 25Hz, let's alternately double and halve again the frequency of this square wave:

If we draw a graph in which up means higher frequency and down means lower frequency, and graph the changes in frequency of our 100Hz square wave, it looks like this:

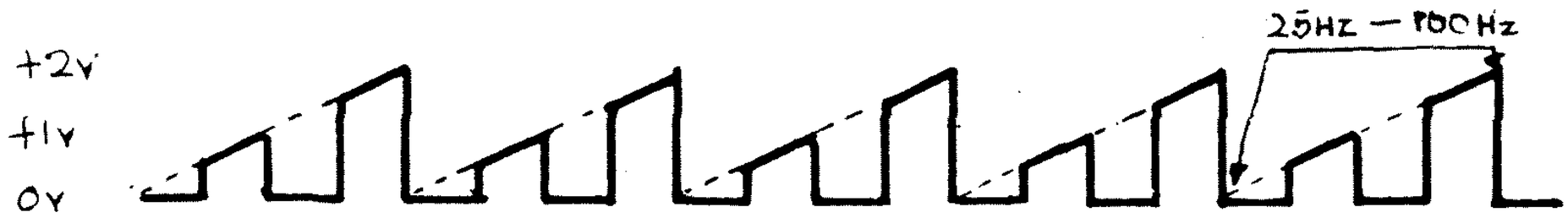


A look at the first and last graphs, and at their "result" in the second graph, shows what we mean if we say that we have FREQUENCY-MODULATED A 100Hz SQUARE WAVE by A 25 Hz SQUARE WAVE.
If, instead of shifting suddenly up and suddenly down again, we start from 100Hz and moved gradually up to say 200Hz and then started suddenly again from 100Hz, a graph of the changes in frequency would look like this:



and in this case we would say we had FREQUENCY-MODULATED A 100Hz SQUARE WAVE WITH A 25Hz SAWTOOTH.
2.1832 AMPLITUDE MODULATION, on the other hand, of a 100Hz square wave by a 25Hz square wave would produce something like this:

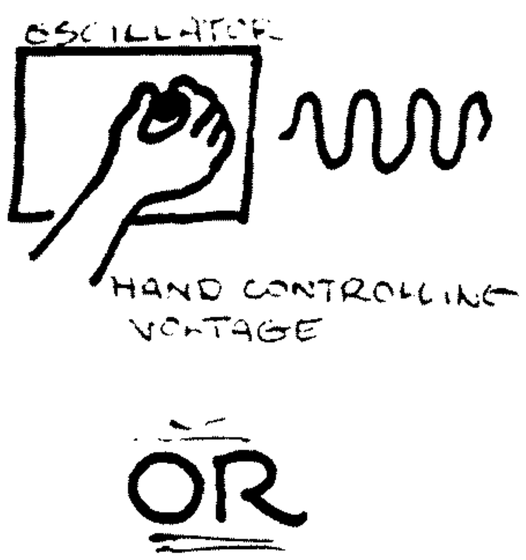
and by a 25Hz sawtooth might look like this:



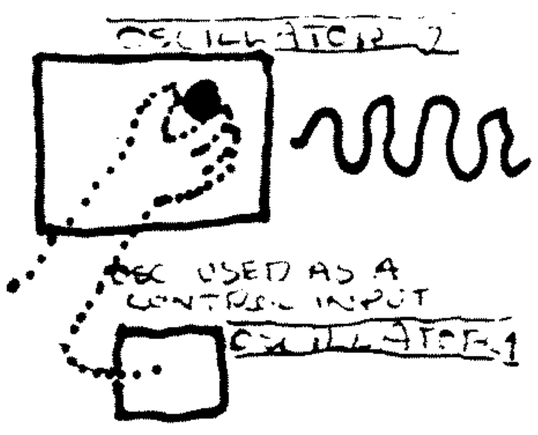
2.2 THE SECOND BASIC IDEA on which electronic synthesis is based is that WAVE-GENERATING and WAVE-MODIFYING equipment MAY BE CONTROLLED ELECTRONICALLY. You may think of this to begin with as a kind of "remote control", in which one piece of equipment may control the frequency or gain of another piece of equipment by an electrical signal.

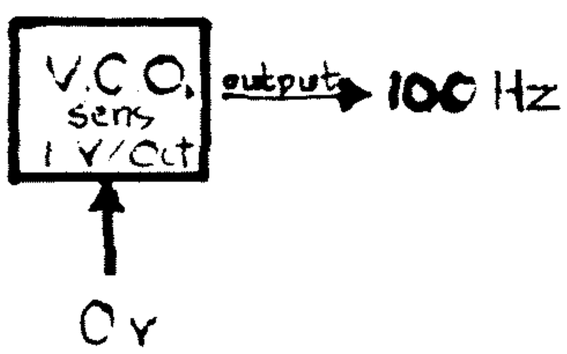
2.201 Other instruments such as electronic organs, electronic pianos, theremins, and so on, generate and/or modify their sounds electronically, just as synthesizers do. But other instruments are designed only for manual control through keys, pedals, stop tabs or knobs, and couplers.

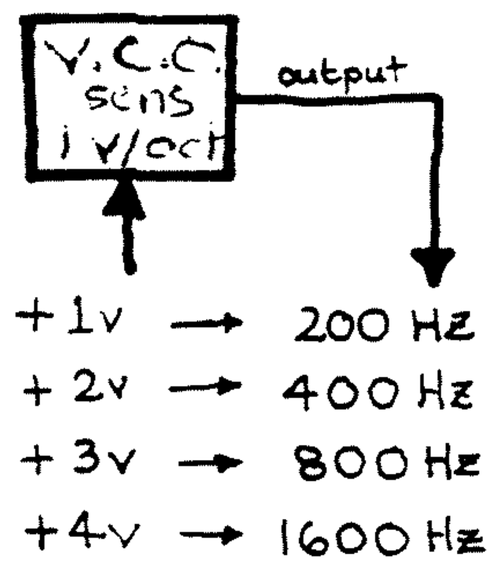
2.21 The oscillators, filters, and amplifiers in a synthesizer, however, are designed to be controlled both manually and by voltages. A vibrato effect, for example, may be obtained by manually varying the frequency of an oscillator rapidly over a small range; but a much smoother and more flexible vibrato is created by using a low-frequency sine wave to control the pitch of the oscillator output.

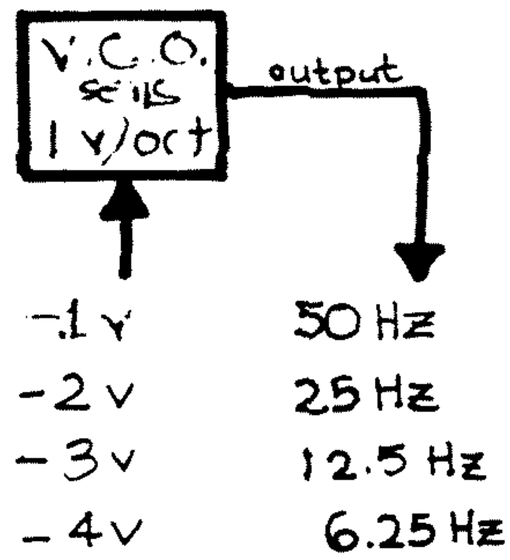


2.211 In any oscillator, filter, or amplifier that is designed to be VOLTAGE CONTROLLED, a voltage applied to a control input will have exactly the same effect as a manual adjustment of one of the controls on the unit. (Which characteristic of the unit is affected is usually indicated at the input to it; some units may be designed so that more than one of their operating characteristics may be VOLTAGE CONTROLLED.) A voltage controlled oscillator, for example, may commonly have a sensitivity of ONE OCTAVE per VOLT. Such an oscillator would double its frequency, i.e. rise one octave, for every increase of one volt in the signal applied to its control input. And a negative one-volt signal at the input would drive the oscillator frequency down by one octave. A continuously fluctuating voltage would cause the oscillator frequency to continuously fluctuate. And so on.

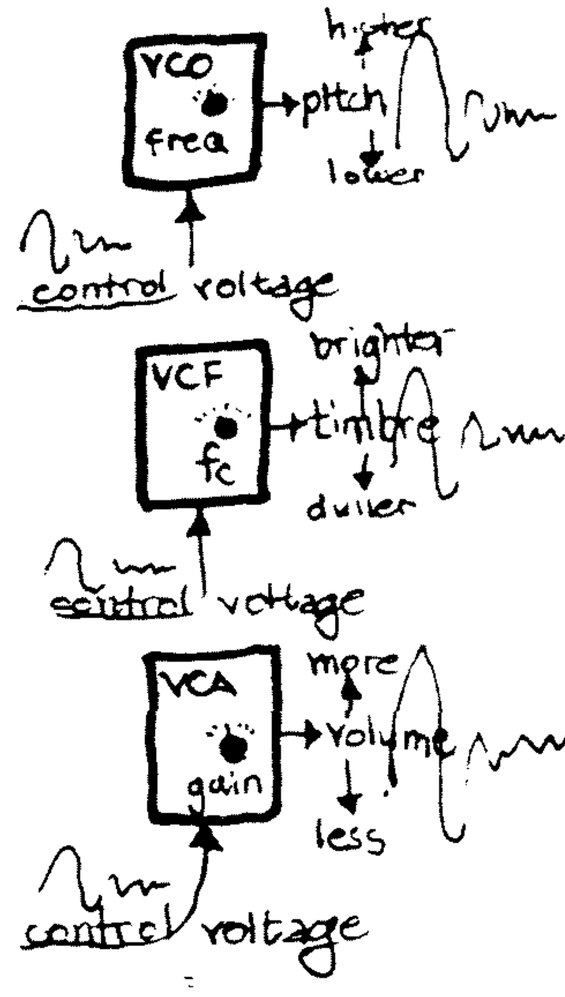








2.212 *It is important to remember that VOLTAGE CONTROL is an* extension of manual control; in theory anything you can do with a voltage you should be able to do with a knob, manually. IN FACT, your hands cannot move fast enough, or accurately enough, to come within miles of what a voltage signal can do with a voltage-controlled device.

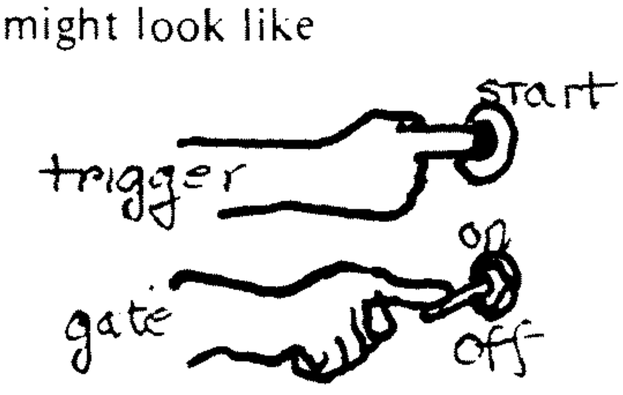


2.22 Continuously variable characteristics such as the frequency of an oscillator, the cutoff frequency of a filter, or the gain of an amplifier, may be controlled by continuously variable voltages.

2.23 Other functions that may be controlled by electrical signals are START functions and ON-OFF functions. Signals used for these purposes are called TRIGGERS and GATES respectively.

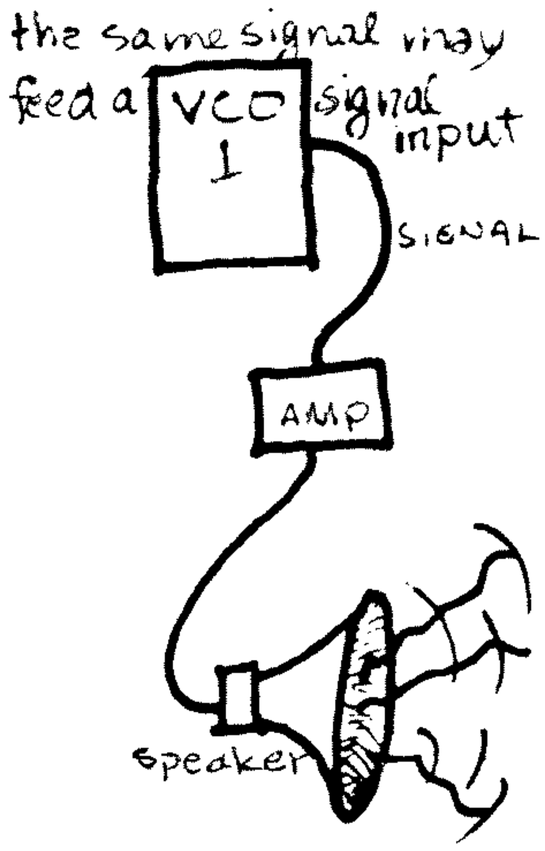
2.231 An electronically TRIGGERED device will begin to operate when a voltage at the proper input goes from zero to some positive voltage and immediately comes back to zero. On a graph such a TRIGGER SIGNAL might look like this ⎍ .

2.232 An electronically GATED device will begin to operate when a voltage at the proper input goes from zero to some positive value AND WILL CONTINUE to operate ONLY AS LONG AS the voltage stays at or above that value. Thus a GATE SIGNAL might look like this ⎍.



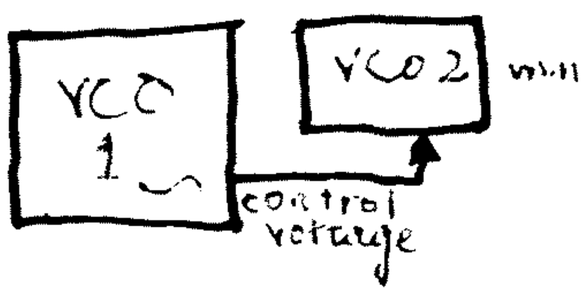
2.24 A keyboard controller for a synthesizer usually works by some combination of CONTROL VOLTAGES to signal which key is pressed down, GATES to signal the length of time a key is down, and TRIGGERS to signal the exact instant at which a key is pressed down.

2.25 USING A SIGNAL TO CONTROL something in this fashion DOES NOT MAKE A DIFFERENCE to the SIGNAL. A voltage is a voltage is a voltage; there are control voltages only where there is equipment designed to be voltage controlled. Calling a signal a control signal is only saying what, in some particular case, its FUNCTION is; namely, that it is being fed to the control input of a voltage-controllable device.



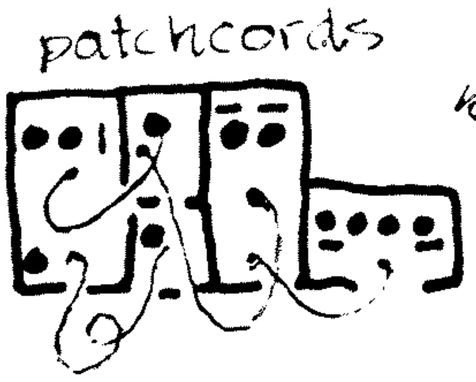
2.251 It is EXTREMELY IMPORTANT to be clear about this because of the OTHER FUNCTION that a voltage signal may serve: i.e. simply to provide an AUDIO SIGNAL (pitched tone or noise) for modification and ultimately for listening. Between these two functions there is all the difference in the world; but there is no intrinsic difference BETWEEN THE SIGNALS THEMSELVES. The same output from the same oscillator may be used either as an AUDIO SIGNAL or as a CONTROL SIGNAL; it makes no difference to the oscillator. To find out whether a signal is audio or control, don't look to where it comes from but rather to where it goes.

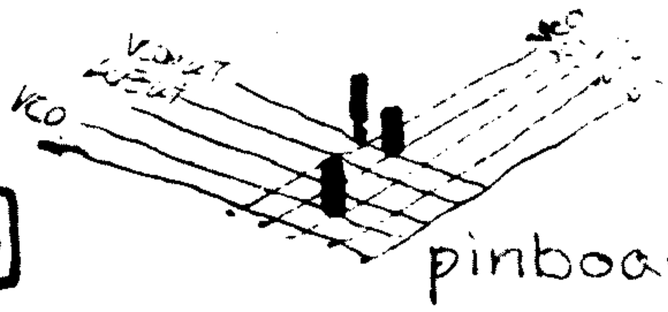
2.3 A SYNTHESIZER, then, consists of SIGNAL-GENERATING and SIGNAL-MODIFYING equipment; and some of this equipment is VOLTAGE-CONTROLLED by signals which usually are generated, and may also be modified, within the synthesizer.



2.31 EACH DEVICE in a synthesizer IS COMPLETELY SEPARATE from the others. It is electrically independent of them and has its own separate INPUT and OUTPUT points, usually readily available to the user.

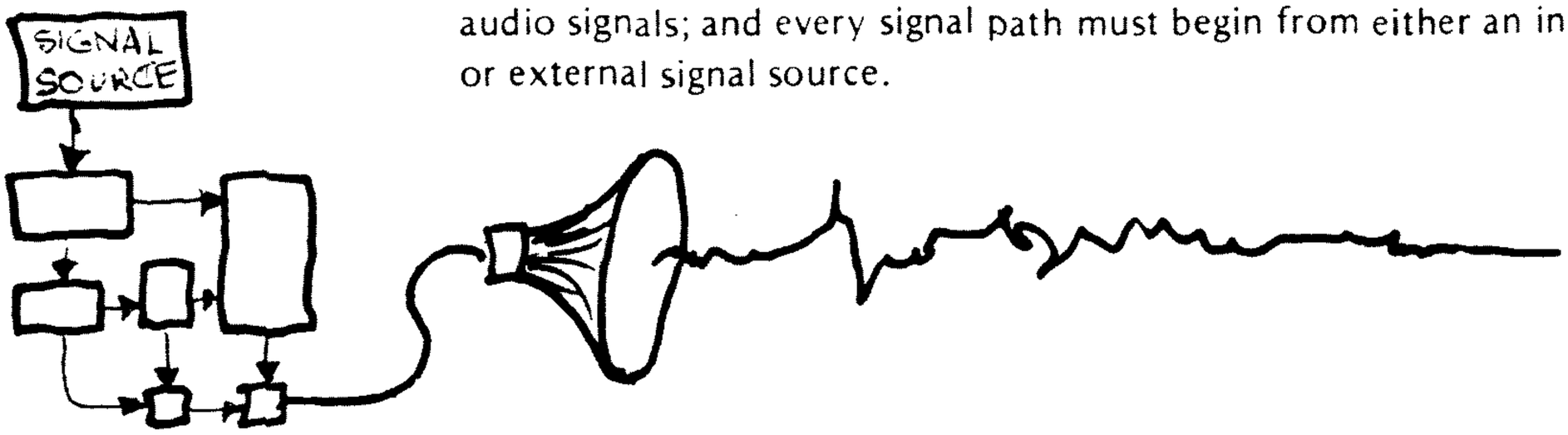
2.32 EVERY SYNTHESIZER has some provision for CONNECTING its internal devices to each other. These may take the form of patch-cords, matrix boards with pins, or matrix switches. The devices are connected to each other BY THE USER, who may connect them in MANY DIFFERENT CONFIGURATIONS depending on what sounds and sound-patterns he wants to produce.





2.33 OPERATING A SYNTHESIZER means doing two different things: one is CONNECTING to each other the devices to be used, or "setting up the patch" as it is called, and the other is SETTING UP, ADJUSTING, AND USING the controls on the devices in the patch.

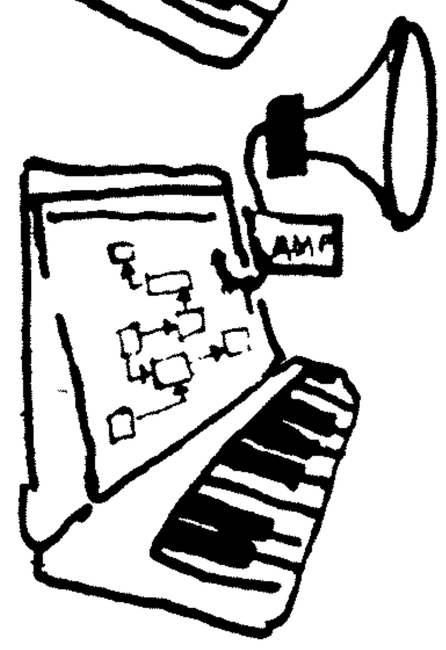
2.331 SETTING UP A PATCH involves at least providing a path for audio signals; and every signal path must begin from either an internal or external signal source.



2.3311 Any signal from any device in a synthesizer may be fed directly to an external amplifier or recorder. More commonly it will be routed through other devices before reaching the synthesizer's main output channels. Several signals may be combined in a mixer, or one signal may be split to follow two different paths, which, modifying it in different ways, may later recombine.

2.3312 Likewise, externally generated signals (from tape recorders, preamplifiers, etc.) may be fed directly into the signal input of any device in a synthesizer, provided only that the signal source output has the proper electrical characteristics (i.e. impedance and amplitude; See Section 4).

2.332 Setting up a patch may also involve providing control paths; but not necessarily.



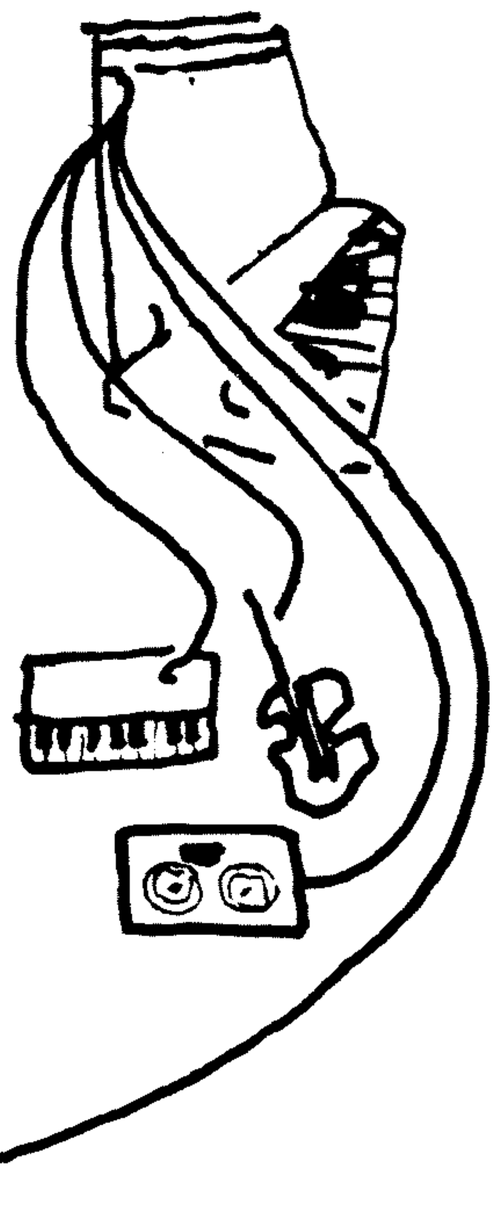
2.3321 It is possible to generate many interesting sounds without ever using voltage control; but it's tedious work, particularly if you're trying to get something like "Mary Had a Little Lamb" or something similar. In the early days of electronic synthesis the only way you could get a tune was either by "playing" an oscillator by hand, as if it were a penny-whistle, or by recording separately each note on tape and then splicing all the notes together. Then for controlling the volume you had to run the completed tape through an amplifier, with one hand on the volume control, while rerecording the result on another recorder. Likewise for controlling the timbre, or tone, of the melody.

2.333 EVERY CONNECTION IN A PATCH IS EITHER A SIGNAL PATH OR A CONTROL PATH.

2.334 NO CONNECTION IN A PATCH CAN BE BOTH A SIGNAL AND A CONTROL PATH AT THE SAME TIME.

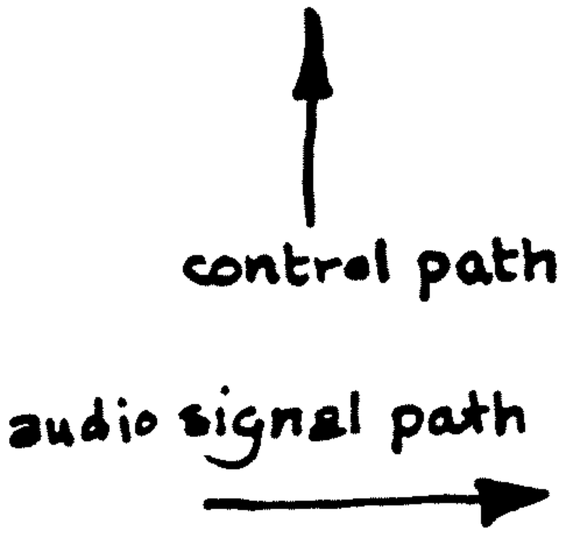
2.335 THE AUDIBLE EFFECT OF MANIPULATING ANY MANUAL CONTROL DEPENDS ON THE PARTICULAR PATCH BEING USED.

2.3351 Thus, for example, if the audio output from an oscillator is part of a signal path, then changing the frequency of the oscillator will in some way change the pitch of a sound in the synthesizer's main output; but if the same oscillator is controlling another oscillator whose output is part of a signal path, then changing the first oscillator's frequency might change the rate of a vibrato or, at a higher frequency, the timbre of a frequency-modulation or amplitude-modulation effect.



2.34 ALL OF THE CONNECTIONS involved in any particular patch can be diagrammed with blocks (to symbolize individual devices) and arrows (to indicate signal and control paths). Such a diagram is called a BLOCK DIAGRAM.

2.341 SIGNAL PATHS in a block diagram are indicated by HORIZONTAL LINES. OUTPUTS leave from the right side of a block, and INPUTS enter from the left.

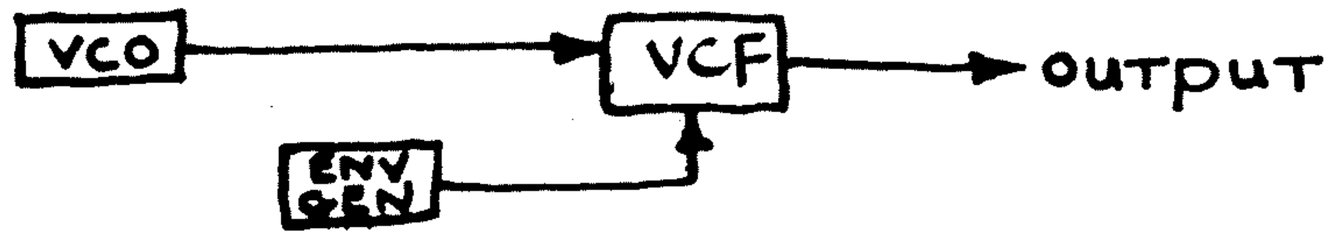


2.342 CONTROL PATHS are indicated by VERTICAL LINES. OUTPUTS leave from the right side of the block and INPUTS enter from the bottom.

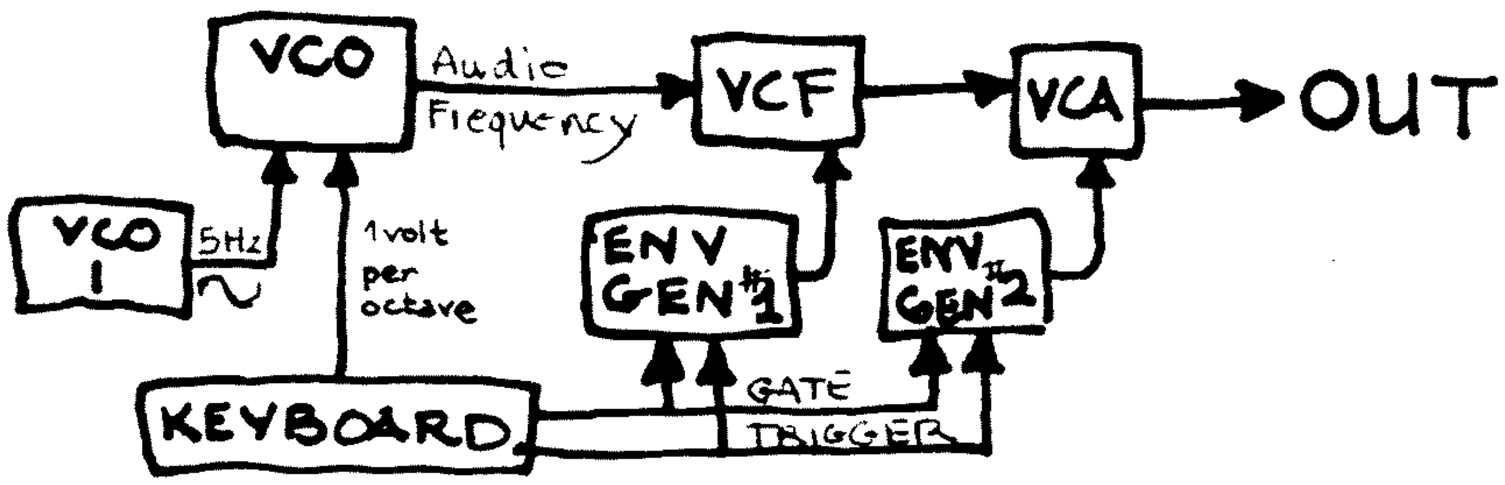
2.343 Inside each block may be specified, if desired, a set of control values for the unit represented by that block:

2.3431 For an oscillator you may want to specify its frequency, or you may want to specify which of several waveforms is being used; for a filter the specification might concern frequency and/or resonance; for a triggered function, the rate of repetition, and so on.

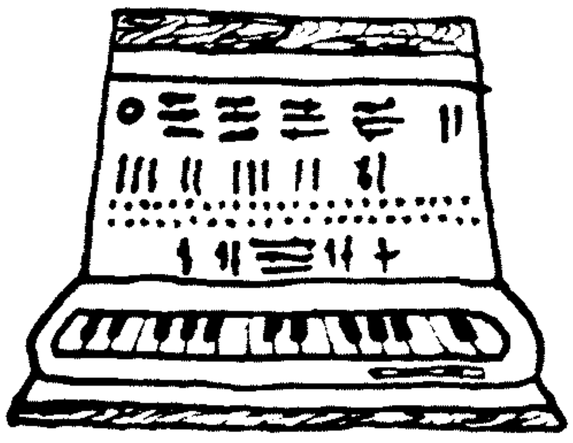
2.344 A simple block diagram might look like this.

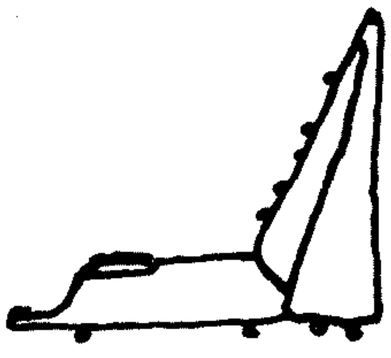


2.345 A more complicated one might look like this.



2.35 BLOCK DIAGRAMS are EXTREMELY USEFUL.





3.0 THE ARP MODEL 2600 SYNTHESIZER is a compact, portable, integrated, dependable, rugged, and supremely flexible synthesizer designed for musicians, teachers, and composers. If you own one, congratulations; if not, commiserations.

3.01 Compact: it measures 31" by 18" by 8½" (main unit) and 31" by 15 3/4" by 3" (3604 keyboard controller).

3.02 Portable: it weighs 36 lbs. naked and 42 lbs. bundled for travel in the accessory carrying case and stand. Main section and keyboard are of about equal weight, with handles for easy, balanced transportation.

3.03 Integrated: non-modular.

3.04 Dependable: it doesn't go out of tune and doesn't change pitch every time you change the patch. Oscillators track accurately over the entire audio spectrum without range-switching.

3.05 Rugged: the 2600 is cased in strong sheet aluminum; all supporting surfaces are protected by rubber feet. Glass-epoxy circuit boards are used throughout, and critical functions are completely encapsulated in separate plug-in units.

3.06 *Supremely flexible: it has three voltage controlled oscillators* with sine, square, sawtooth, triangle, and variable-width pulse outputs; a random noise generator, voltage-controlled lowpass filter and resonator, two envelope generators, a voltage-controlled amplifier, ring modulator AC or DC coupled, envelope follower, microphone preamp with 20-40-or-60 Db gain, reverb, a signal inverter, another one, a lag voltage processor, an electronic switch and clock, sample and hold circuit, mixer, pan, and four-octave keyboard with portamento and interval squeezer. And monitor speakers and a headphone jack. For stereo headphones.

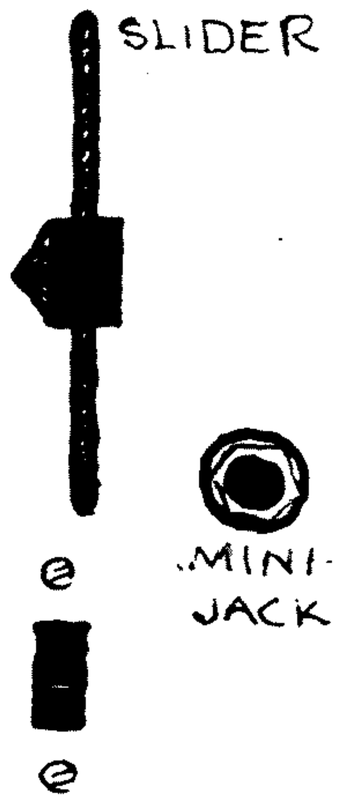
3.07 FOR MUSICIANS INTERESTED IN LIVE PERFORMANCE, the Model 2600 features a built-in patch that makes most popular synthesizer effects available without even touching a patchcord. This internally "hard-wired" patch may be overridden, however, by inserting a patchcord plug into the appropriate panel jack. The stable oscillators may not need tuning for weeks at a time. And the envelope follower (see 4.33) allows the 2600 to create spectacular interactions among other electric instruments connected to it.

3.08 FOR TEACHING PURPOSES, we have designed the panel of the 2600 to be as clear as possible. All signal and control paths are indicated by arrows. The built-in convenience patch mentioned in 3.1 may be entirely disregarded; circuits and functions may be demonstrated one at a time or in any combination whatever. Each function is clearly labelled in a box reminiscent of "block diagrams", and the signal paths to and from voltage-controlled units are consistent with a set of rules for block diagramming that we have given in 2.34 through 2.35. The 2600 *outputs from all functions are compatible with external teaching* aids such as oscilloscopes.

3.09 A modest additional investment in tape recorders, equalizers, and noise reduction equipment will put any composer of electronic or concrete music inside A COMPLETE STUDIO FOR TAPE COMPOSITION. Prerecorded signals may be routed through any individual function of the 2600 for partial transformation or may become part of longer signal paths. The envelope follower generates a control voltage from any signal fed into it, and makes possible the modulation of one signal by the envelope of another.

3.1 IF YOUR FIRST GLIMPSE of the ARP Model 2600 is somewhat confusing, look again. The control-panel graphics are as self-explanatory as we could possibly make them, and some careful study before plugging it in and turning it on will more than repay the time it takes to do so. The Model 2600 consists of two units; one is the synthesizer itself and the other is the keyboard controller 3604.

3.11 Concentrate first on the synthesizer itself. Your only concern is the various objects protruding through the front of it. Notice (if you hadn't already) that these are mostly either SLIDER CONTROLS or MINI-JACKS. Besides these, there are only seven slide switches, one button (it's very small), one stereo headphone jack, and one knob. (A number of semi-fixed controls are recessed behind the panel. They are dealt with in Section 7, and should not need adjustment for long periods of time; your 2600 was thoroughly calibrated and checked before leaving the factory. DO NOT TOUCH THEM until you are COMPLETELY FAMILIAR WITH YOUR INSTRUMENT and with the contents of Section 7).



3.111 THERE ARE EIGHTY-ONE mini-jacks on the front panel. FORTY-FIVE are unequivocally INPUTS and TWENTY-NINE are unequivocally OUTPUTS.

45
29
74

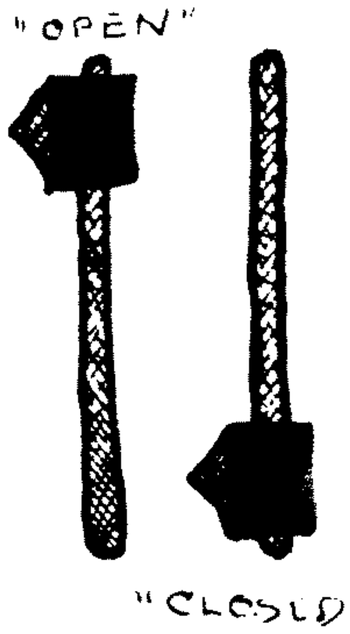
The remaining seven jacks belong to the ELECTRONIC SWITCH and the MULTIPLE OUTLET. Because the switch works in either direction, it has either two inputs and one output or one input and two outputs depending on which direction you're thinking of; the MULTIPLE OUTPUT distributes one input to 1, 2, or 3 outputs.

3.112 OF THE 45 UNEQUIVOCAL INPUTS, 32 are all in a row running across the center of the panel. (There are 34 jacks in the row; the two labelled "gate" and "trig" are outputs.) This row of inputs divides the control surface almost evenly in half. ABOVE THIS ROW, in the upper half of the control surface, THERE ARE ONLY FOUR INPUT JACKS. One of them is the input to the microphone preamp in the extreme upper left corner. The other three are at the upper right corner, labelled LEFT INPUT, PAN, and RIGHT INPUT respectively. ALL OF THE OTHER JACKS IN THE UPPER HALF OF THE CONTROL SURFACE ARE OUTPUTS. In the lower half of the control surface, the jacks numbered 1 through 7 are inputs to the voltage processor, and of the column of four jacks in the section labelled "sample and hold", the upper and lower jacks are inputs.

3.113 All the remaining jacks are outputs. Most of them are labelled as such; a few are not but have arrows pointing to them, e.g. in the voltage processor, the three jacks furthest to the right are outputs; and in the envelope generator section of the upper half of the panel, the two jacks labelled gate and trigger are outputs.

3.114 THERE ARE FIFTY-SEVEN SLIDER CONTROLS.

3.1141 THIRTY-FIVE OF THESE are ATTENUATORS. And of the 35 attenuators, 26 are in a row just above the row of jacks. (There are 28 slider controls in the row; the two on either side of the box labelled ATTACK RELEASE are not attenuators.)



3.11411 An ATTENUATOR is the OPPOSITE of an AMPLIFIER. And much simpler too. An attenuator "wide open" or all the way up passes

a signal through at full strength. "Closing" the attenuator control slowly—i.e. sliding it down—gradually weakens the signal passing through it until there is no signal at all.

3.11412 MOST OF THE ATTENUATORS in the 2600 are directly associated with either an input to something or an output from something. Each of the vertical attenuators across the middle of the 2600, for example, controls the amplitude of a signal coming through the input directly below it.

3.1142 There is no one thing that the other slider controls do; the function of each one is indicated separately on the panel.

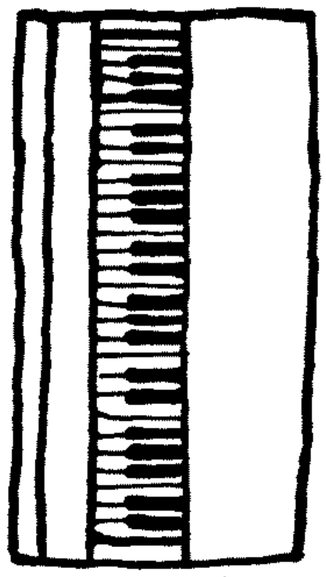
3.115 The knob is a rotary attenuator for signals fed to the microphone preamplifier

3.116 The button is a manual starter for the envelope generators.

3.117 One of the switches is a power switch to turn the 2600 on and off. Of the other six, one selects three ranges of amplification in the mike preamp, one selects AC or DC coupling of the ring modulator, three select frequency ranges for the VCO's, and one selects keyboard or other control of the envelope generators.

3.118 The headphone jack is for a standard stereo headphone plug. Inserting one disconnects the speakers. The headphones should be 8-ohms impedance.

3.119 That's all there is to the main unit.



3.12 The 3604 keyboard has four octaves and is plugged into the main unit by two short cables, one at each end. The plug blades are arranged so that it is impossible to plug them in the wrong way.

3.121 In addition to keys—49 of them—the keyboard unit has three switches and three knobs. The switches are on-off and the knobs are more-or-less.

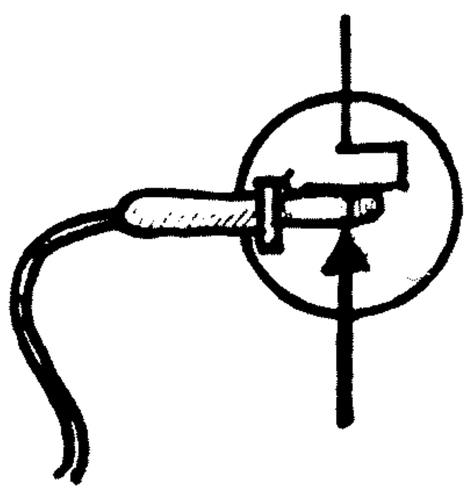
3.2 THE MOST USEFUL and MOST FREQUENTLY USED connections in the Model 2600 ARE PREWIRED.

3.21 To see a simple example of this, note that the first input jack on the left, in the row of jacks running across the middle of the 2600, is an input to the ENVELOPE FOLLOWER. Underneath the jack is a symbol that looks like this

and under the symbol is a

small box labelled MIC PRE-AMP. The symbol is simply a diagram of the construction of the mini-jack itself: when a plug is inserted this is what happens. THE OUTPUT FROM THE MICROPHONE PREAMP IS PREWIRED TO THE ENVELOPE FOLLOWER INPUT, AND IS AUTOMATICALLY DISCONNECTED WHENEVER A PLUG IS INSERTED INTO THE JACK INPUT TO THE ENVELOPE FOLLOWER.

3.22 For another simple example, note that in the same row of jacks, the third one from the right is an input to the MIXER, and that it is prewired to the output from the VOLTAGE-CONTROLLED FILTER (VCF). Note again that the fifth of the five AUDIO INPUTS to the VCF carries a prewired signal from the NOISE GENERATOR (counting across from left to right this is the 21st jack). You can listen to this input by raising the LEFT and RIGHT SPEAKER ATTENUATORS, the VCF input to the MIXER, and the NOISE GEN input attenuator to the VCF, each about halfway (make sure that ALL the other attenuators are DOWN). Now experimenting with the two horizontal control sliders at the top of the VCF panel will give you a wide range of filtered sounds. (With the RESONANCE CONTROL at MAXIMUM the VCF output is a pure tone independent of any input. In this condition the VCF is OSCILLATING and its output is a SINE WAVE.)



3.23 It will be worth your while to experiment thoroughly and systematically with the built-in patch connections at this point, particularly if you are planning to use the 2600 as a live performance instrument. In section 4 we describe the main functions of each separate unit in the 2600, and in section 5 we give sample patches for further experimentation; for the present we will only give you a few general principles to keep you from going out of your skull with complications.

3.231 EXPERIMENT WITH ONE SIGNAL AT A TIME. With the VCF, for example, when you have listened to everything the filter can do with a noise input, close that input all the way down and use the VCO-3 SAWTOOTH input immediately to its left. Now you can experiment not only with the VCF controls, but also with the manual FREQUENCY controls of VCO-3; and when you have done that, experiment one by one with the CONTROL INPUT signals to VCO-3.

3.232 DO NOT, for the present, USE THE KEYBOARD. For now, your experiments should be directed toward familiarizing yourself with the SOUND-GENERATING CAPABILITIES of the 2600. A synthesizer keyboard is MISLEADINGLY FAMILIAR and doesn't really operate the way any other keyboard does, and you'll find out soon enough just how it does operate.

3.2321 The 3604 keyboard unit works something like the frets on a rubber guitar, turned inside out.

3.233 In making use of the ADSR and AR control inputs to anything, set the SWITCH just under the ATTACK-RELEASE box to its UPPER position. Then use the MANUAL START button to initiate an envelope output from either the ADSR envelope generator or the AR generator. Do anything: punch it, hold it down, or tap it repeatedly.

3.234 Follow prewired connections BACKWARDS to their sources. Many of the VCO outputs, for example, are not prewired to any input at all, and it would be a waste of time trying to find out "where it all goes". But most of the input jacks do have a prewired connection which you can trace back from the boxes and labels just beneath them.

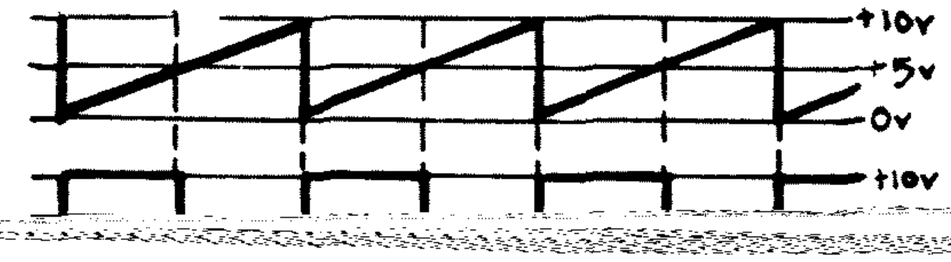
3.235 For every effect you hear, ask yourself which 2600 function is creating which part of the effect. It will help to draw or imagine block diagrams for each sound.

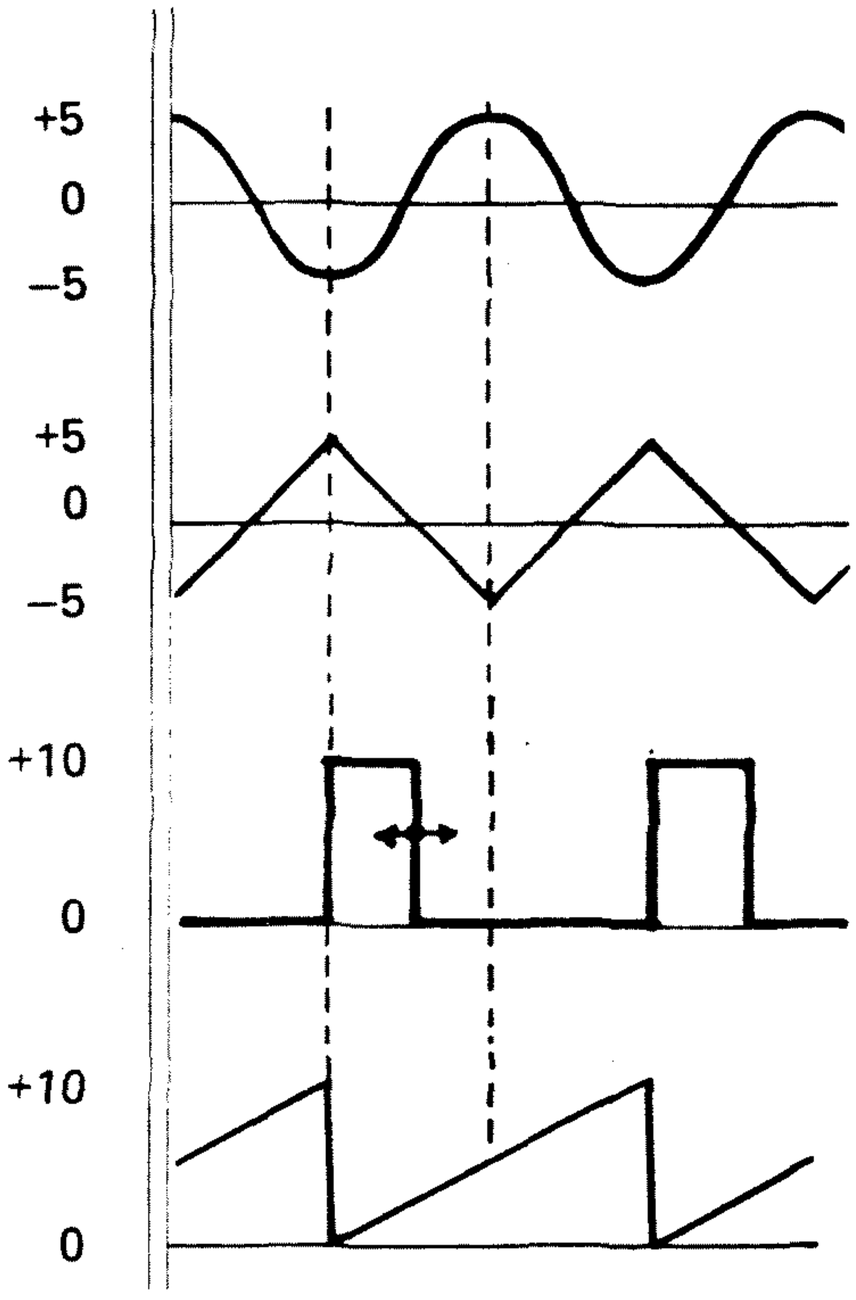
4.0 IN THIS SECTION we will discuss in detail the operating characteristics of each separate function of the Model 2600.
4.01 BY THIS TIME you should have experimented in some measure with the operation of the 2600 through its built-in patch, and you should therefore have some idea of the enormous variety of sounds the 2600 can generate.
4.02 MORE IMPORTANT, HOWEVER, is the fact that you are beginning to get an idea of how the individual functions can operate on each other to control the audio output from the 2600.
4.03 That is important because throughout this section we are going to describe each function mainly in terms of what it does to WAVEFORMS—NOT in terms of what "sounds" it makes, or what it does to "sounds". You should be able to understand why such an approach would be hopelessly complicated: refresh your memory if necessary by rereading 2.335 and 2.3351.
4.04 Think of this section as a REFERENCE section. As you gain familiarity with the 2600, and particularly as you gain experience in translating from the language of SOUNDS to the language of WAVEFORMS and back again, you will find that this section will become more and more valuable to you. For THE MORE CLEARLY YOU UNDERSTAND each function PURELY IN TERMS OF WAVEFORMS, the more clearly will you see the possibilities for NEW PATCHES and NEW USES for each function in the 2600.
4.1 Five of the 2600 functions are voltage controlled: these are the three VCO's, the VCF, and the VCA.

4.11 The three VOLTAGE CONTROLLED OSCILLATORS share some common characteristics.

1. Each VCO has two operating ranges selected by a switch in its lower left corner. The range labelled AUDIO is from 10Hz to 10KHz, and the range labelled LF (for LOW FREQUENCY) is from .03Hz, or one cycle approximately every 30 seconds, to 30Hz.
2. Within either operating range, the frequency of each VCO is manually controlled by two sliders. One is labelled INITIAL OSCILLATOR FREQUENCY and covers about ten octaves. The other is labelled FINE TUNE and covers a little less than half an octave.
3. Each VCO accepts up to four frequency-control inputs (labelled FM INPUTS). Three of these are governed by logarithmic input attenuators; one is unattenuated and is prewired in each VCO to the KEYBOARD CONTROL VOLTAGE.
4. Use of any VCO in its LF range automatically disconnects it from the KEYBOARD CONTROL VOLTAGE. If it is desired to reinstate this connection during LF operation this must be done by patch-cord.
5. The sensitivity of each VCO to control voltages is 1V/Octave. The impedance of the control inputs is 50Kohm minimum. The signal output impedance is 1Kohm.

4.111 The first oscillator, VCO-1, generates SAWTOOTH and SQUARE waves at two independent outputs. Peak amplitude of each signal is +10V; phase relationship is given in the dia-

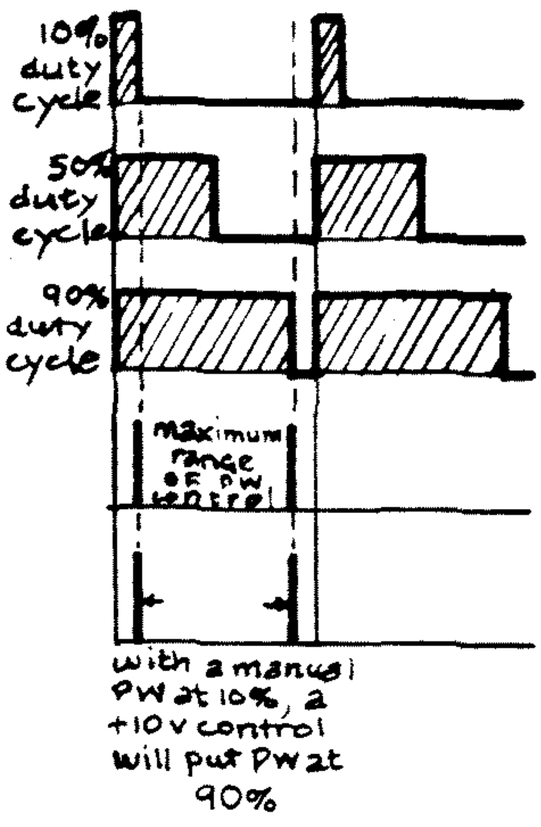




4.112 The second oscillator, VCO-2, generates SINE, SAWTOOTH, TRIANGLE, and variable width PULSE outputs at four independent outputs. Amplitude and phase relationships are given in the diagram at the left.

Note that the SAWTOOTH and PULSE outputs are positive-going only, while the TRIANGLE and SINE waveforms are balanced on either side of OV. When these waveforms are used to control the frequency of one of the other VCO's, for example, the sawtooth and pulse outputs will only be capable in general of raising the VCO frequency, while the triangle and sine outputs will alternately raise and lower the VCO frequency relative to its initial frequency setting.

Note too that the triangle and sine wave outputs are out of phase; where one is positive the other is negative. Adding them directly together would simply cancel the fundamental sine-wave component in the triangle waveform and leave a waveform representing only the upper harmonics.

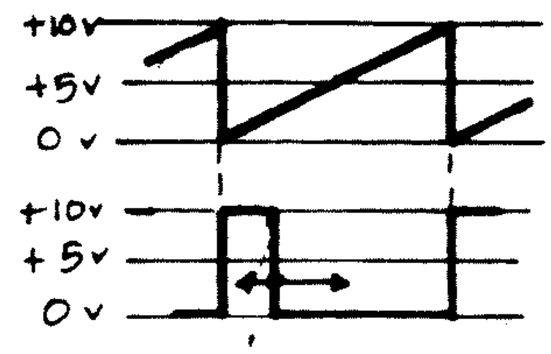


Outputs from the Sample and Hold circuit (S/H OUT), ADSR Envelope Generator, and VCO-1 SQUARE output are prewired to the three attenuator-governed frequency-control inputs. These connections may be interrupted by insertion of patch-cord plugs or dummy plugs into the input jacks. The Keyboard Control Voltage, normally prewired to the unattenuated control input, is automatically disconnected either by insertion of a patch cord into the input or by use of the oscillator in its LF range. In the latter case, if for any reason the keyboard voltage is needed it may be patched in.

PULSE WIDTH is manually controllable by a slider (located just under the two sliders for manual control of frequency) over a range from 10% to 90% duty cycle. Pulse width is also subject to voltage control; the sensitivity is 10% of range/volt. There is one attenuator-gov-

erned input for control of pulse width; it is labelled PULSE WIDTH MODULATION. The Noise Generator output is prewired to this input.

4.113 VCO-3 generates SAWTOOTH and variable-width PULSE outputs simultaneously. Amplitude and phase relationships are given by the diagram in the margin.



Pulse width is manually controlled by a third slider under the two frequency-control sliders and is variable from 10% to 90% duty cycle.

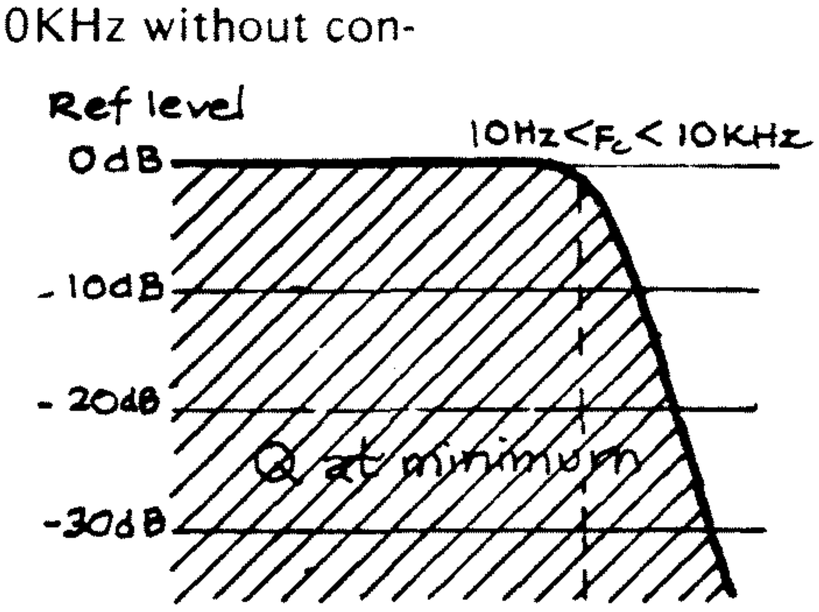
Signals from the Noise Generator, the ADSR Envelope Generator, and VCO-2 Sine Wave are prewired to the three attenuator-governed control inputs. The Keyboard Control Voltage, normally prewired to the unattenuated control input, is automatically disconnected either by insertion of a patch cord into that input or by use of the oscillator in its LF range. In the latter case if for any reason the keyboard signal is desired for control purposes it may be patched in.

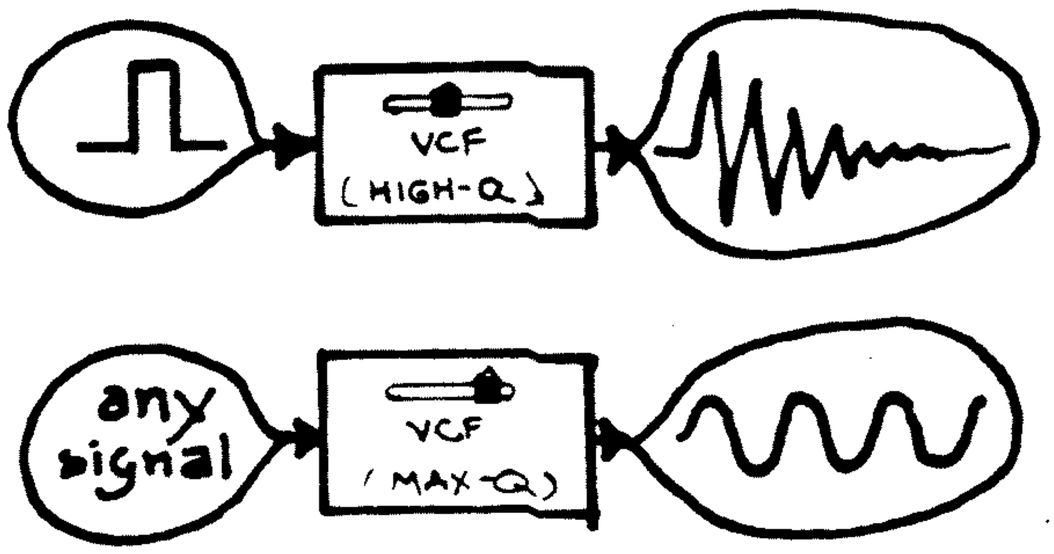
4.12 The VOLTAGE CONTROLLED FILTER/RESONATOR is a low-pass filter with variable cutoff frequency (F_C) and resonance (Q). The response below F_C is flat down to DC; above F_C the response falls off at 24Db per octave. F_C range is from 10Hz to 10KHz without control voltages; under voltage control F_C can be driven as far down as 1Hz and as high as 20KHz.

F_C is controlled manually by a COARSE tuning slider (labelled INITIAL FILTER FREQUENCY) and a FINE TUNE slider. F_C may also be controlled by external voltages; the sensitivity under voltage control is 1V/Octave.

The Q, or resonance, of the filter circuit is controlled by a single manual slider. As the Q is increased by moving this slider from left to right, the response below F_C is gradually attenuated until a sharp peak remains at the cutoff frequency. (Gain at F_C is always unity.)

At this setting, just below the point at which oscillation begins, the filter will ring distinctly in response to any sharply defined pulse presented to its signal input. In this state it is a precise electronic analogue of any highly resonant physical





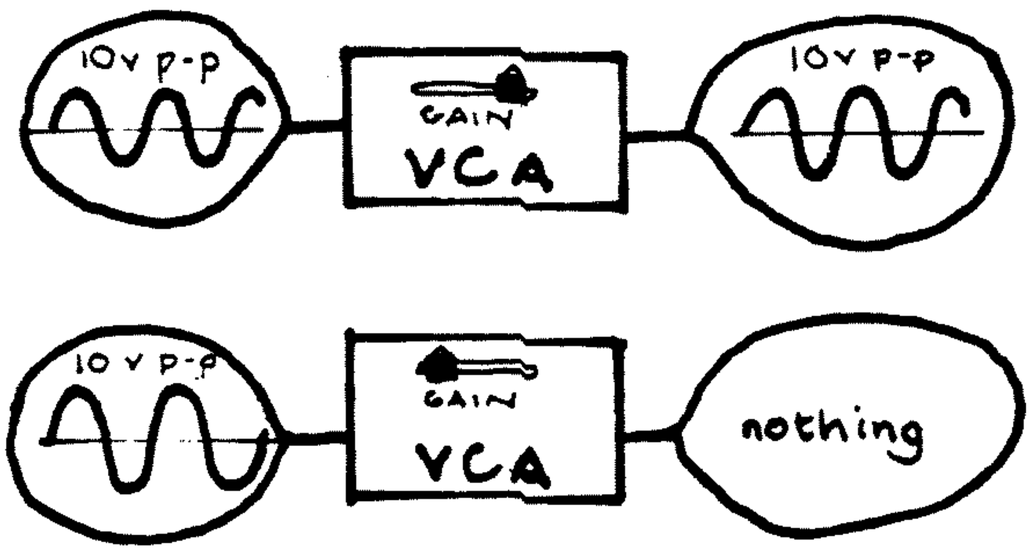
system and may be used for various percussion effects depending on its resonant frequency (identical to F_C) and on the pulse or train of pulses exciting it.

As the Q is raised still higher, beyond about the halfway point in the control slider travel, the filter will begin to oscillate and will produce a pure sine-wave output even in the absence of any signal input.

4.121 There are FIVE SIGNAL INPUTS to the VCF. They are fed through logarithmic attenuators to a summing point, and then to the VCF itself. Prewired connections are from the Ring Modulator, VCO-1 Square, VCO-2 Pulse, VCO-3 Sawtooth, and Noise Generator outputs. Any of these is automatically disconnected by insertion of a patch cord into the proper input jack. Each input has an impedance of not less than 50Kohms.

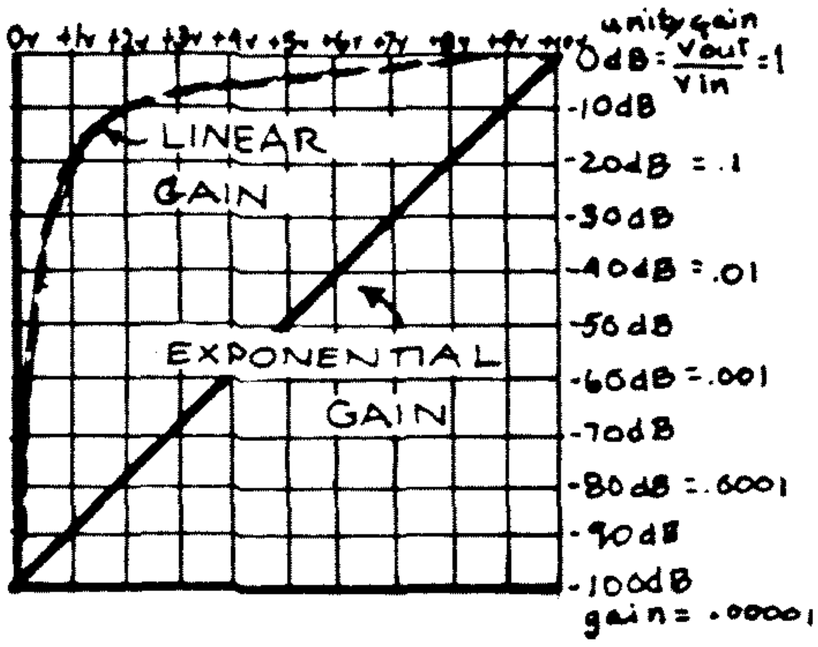
There are THREE CONTROL INPUTS to the VCF. Two are fed through linear attenuators; prewired to these are the ADSR Envelope Generator output and the VCO-2 Sine output. The third, unattenuated, is normally from the Keyboard Control Voltage. Any of these is automatically disconnected by insertion of patch cords into the proper input jack. The impedance of the control inputs is not less than 50Kohms.

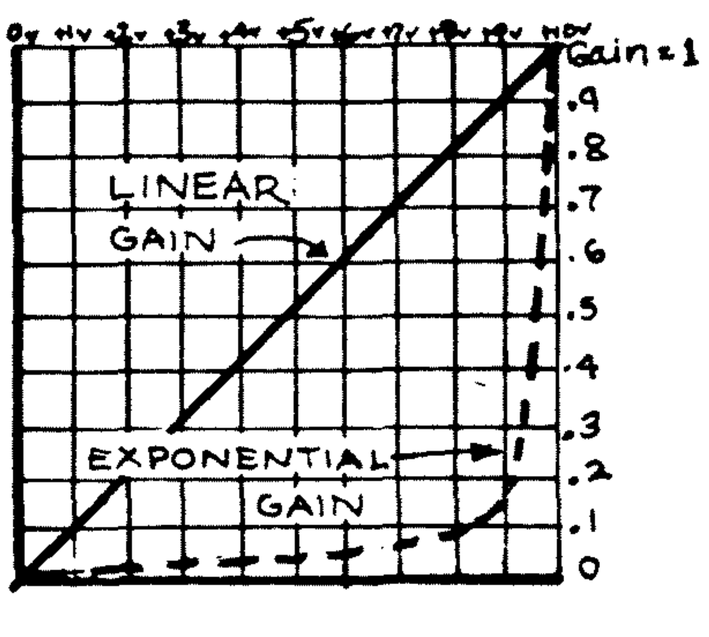
4.122 The output impedance of the VCF is 1Kohm and the signal-to noise ratio is greater than 60Db referred to a 10V P-to-P input.



4.13 The VOLTAGE CONTROLLED AMPLIFIER has a maximum gain of unity and a dynamic range of 100Db. With the INITIAL GAIN control at maximum, and with no control input, the VCA will pass with unchanged amplitude any signal presented to its signal input. On the other hand, with the INITIAL GAIN control at minimum, no signal will pass through the amplifier at all unless some positive voltage is present at one or both of its control-voltage inputs.

One of these inputs has a LINEAR sensitivity. The gain of the amplifier in response to a control voltage presented through this input will equal $V_C/10$, i.e. dividing the value of the control voltage by 10 will





give the gain factor.

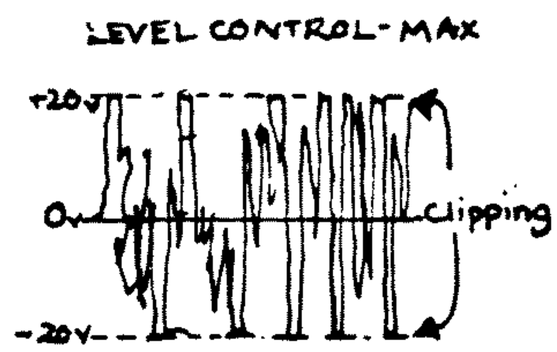
The other control input has an EXPONENTIAL sensitivity. The gain of the amplifier in response to a control voltage presented through this input will equal 10Db/Volt.

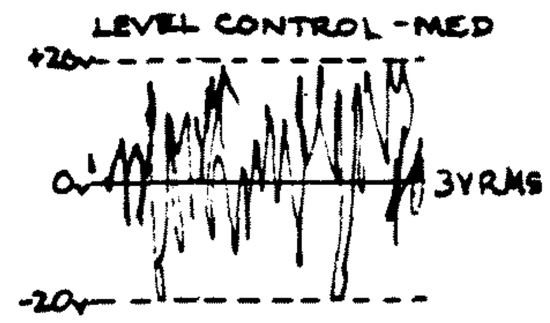
4.131 There are TWO SIGNAL INPUTS to the VCA. Prewired connections to these are from the VCF and the Ring Modulator. Either of these may be automatically disconnected by insertion of a patch cord into the proper input jack.

Prewired to the LINEAR control input is the output from the AR Envelope Generator, and to the EXPONENTIAL control input goes a prewired connection from the ADSR Envelope Generator output. Either is automatically disconnected by insertion of a patch cord into the proper input jack.

Signal and control input impedances are not less than 50Kohms. Signal output impedance is 1Kohm. The signal-to-noise ratio is greater than 60Db referred to a 10V P-P input signal.

4.2 The NOISE GENERATOR produces an audio noise signal of at least ten octaves bandwidth. Manual controls govern LEVEL and COLOR.





The LEVEL control, at minimum, cuts off the output signal completely. At maximum, the output is clipped at 20V P-P to produce binary, or two-valued, noise. Clipping begins with the level control approximately half open. The maximum RMS output, if clipping is to occur less than 1% of the time, is 3V.

The COLOR control is continuously adjustable from WHITE noise to LOW-FREQUENCY noise output. In the latter case the output falls off at the rate of 6Db/Octave; the PINK noise position approximates a 3Db/Octave slope.

There is one signal output. Impedance is 1Kohm.

4.3 Several of the 2600 functions are used almost exclusively for control purposes, although each of them may under certain conditions generate output waveforms at audio frequencies. They are not in any

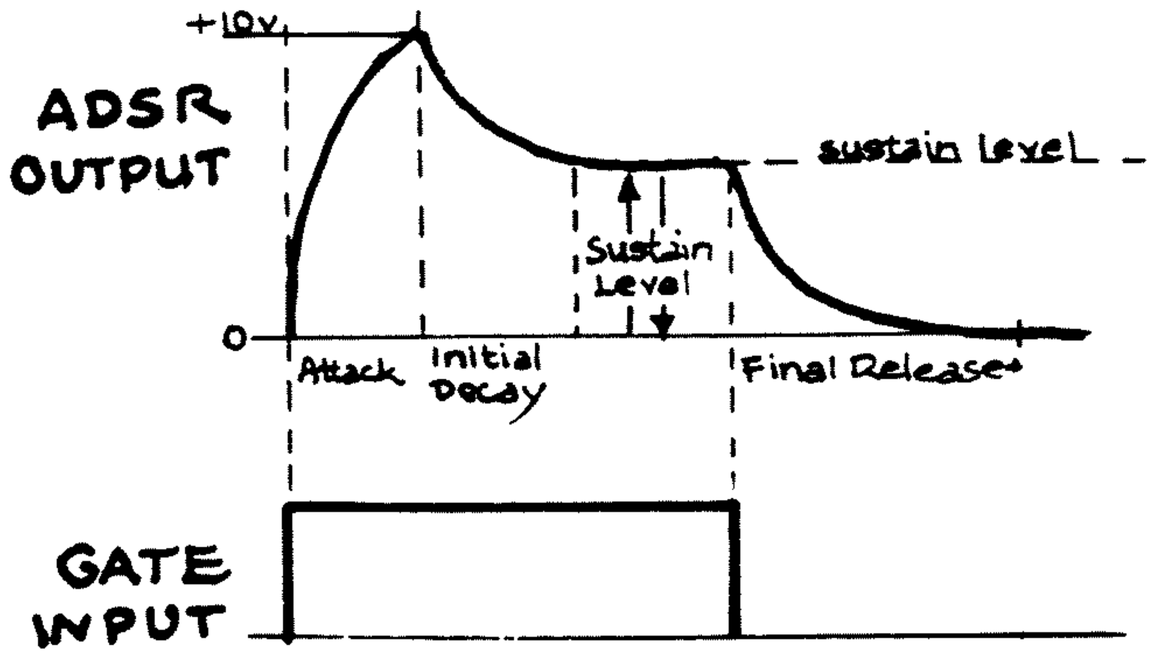
way damaged by such operation (a voltage is a voltage is a voltage), but it is the exception and not the rule. They are therefore grouped here as CONTROL-SIGNAL GENERATORS.

4.31 The two ENVELOPE GENERATORS produce transient waveforms with controllable rise and fall times, and are used primarily with the VCF and FCA in generating events whose fluctuating harmonic and amplitude characteristics must be accurately and repeatably controlled. Note that the output from both generators is in itself merely a positive-going voltage whose rise and fall time is governed by the slide controls on the generator itself and whose onset and overall duration is governed in turn by a GATE voltage which controls the generator.

The maximum value that either envelope can reach is +10V; thus, unattenuated, either envelope is capable of driving a VCF or VCA from its minimum initial setting (10Hz for the VCF, –100Db for the VCA) all the way up to maximum. See 4.12 and 4.13, specifically the data on control input sensitivity. Reread too sections 2.16 through 2.17.

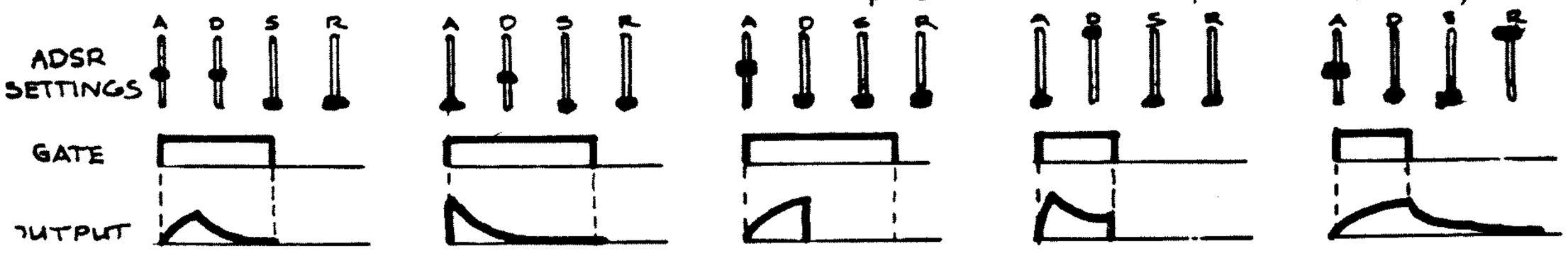
GATES for the operation of the envelope generators may be provided by a MANUAL START button, the 3604 keyboard controller, or any +10V square-wave or pulse signal presented to the external input indicated by the panel graphics. The two latter sources are switch-selected by the slide switch just under the lower (A/R) generator. The MANUAL START button overrides both of these.

The output impedance of both generators is 1Kohm.



4.311 The upper ENVELOPE GENERATOR offers controllable ATTACK time, INITIAL DECAY time, SUSTAIN level, and FINAL RELEASE time. Four vertical sliders control these four parameters: note that three of these are time parameters and the fourth—sustain level—is not.

The generator produces an output only when a GATE signal is present at its input. At the onset of a gate signal, the output voltage rises to +10V at a rate determined by the setting of the ATTACK time control. If the gate is still present when the output reaches +10V, the out-



put immediately begins to decrease, at a rate determined by the setting of the INITIAL DECAY control, to a level governed by the SUSTAIN level control. The output voltage will remain at this level as long as the GATE signal remains at the generator input. When the gate signal is removed, the output voltage returns to zero at a rate determined by the FINAL RELEASE time control.

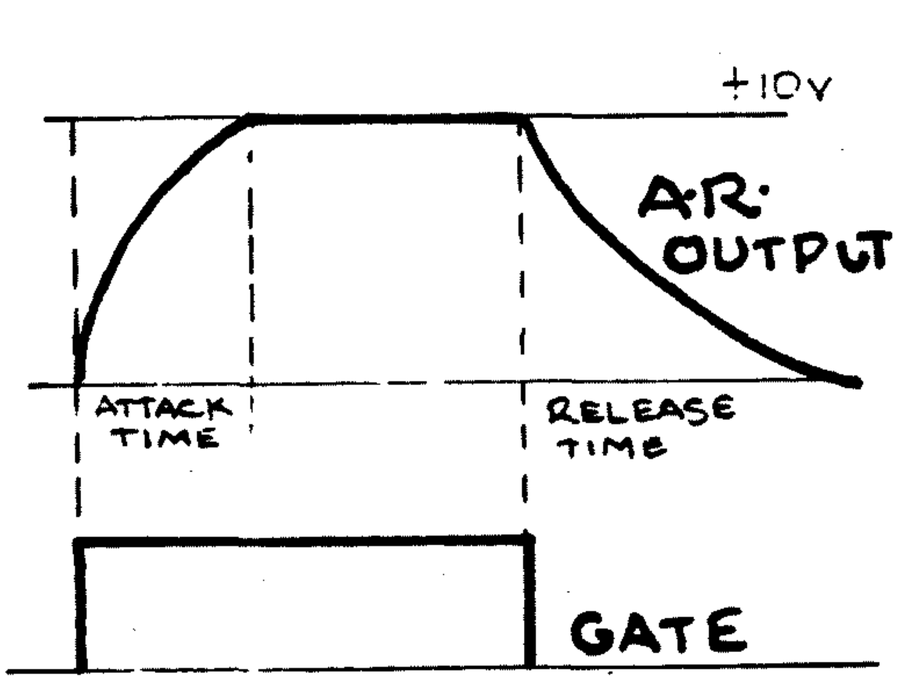
If the GATE signal is removed at any point during the envelope, the output voltage immediately begins to decrease to zero at a rate determined by the setting of the FINAL RELEASE time control.

When the envelope generator is controlled from the 3604 keyboard controller, it responds to the TRIGGER signal from the keyboard—supposing that an envelope is already in process—by repeating the first two stages of an envelope and returning again to the SUSTAIN level. For all other control gates presented to the generator input, circuitry internal to the generator itself derives a TRIGGER signal from the leading edge of the gate.

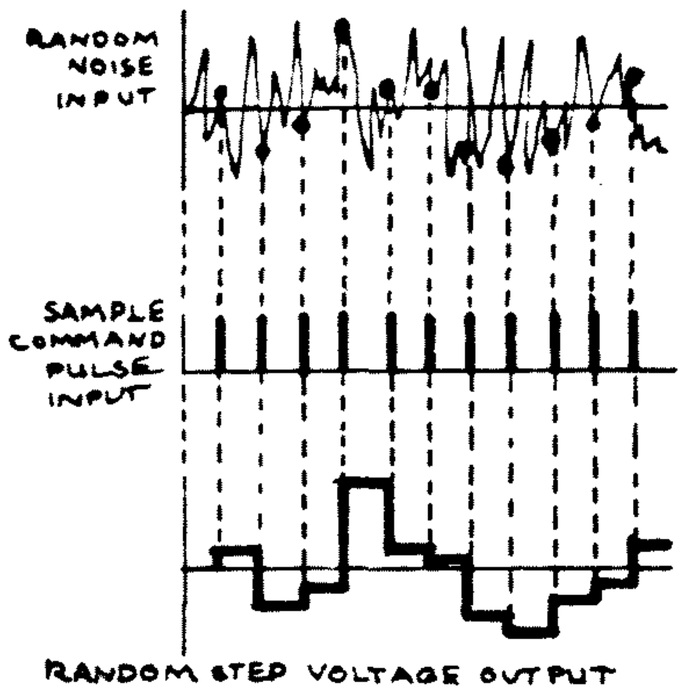
4.312 The lower ENVELOPE GENERATOR offers controllable ATTACK and RELEASE time.

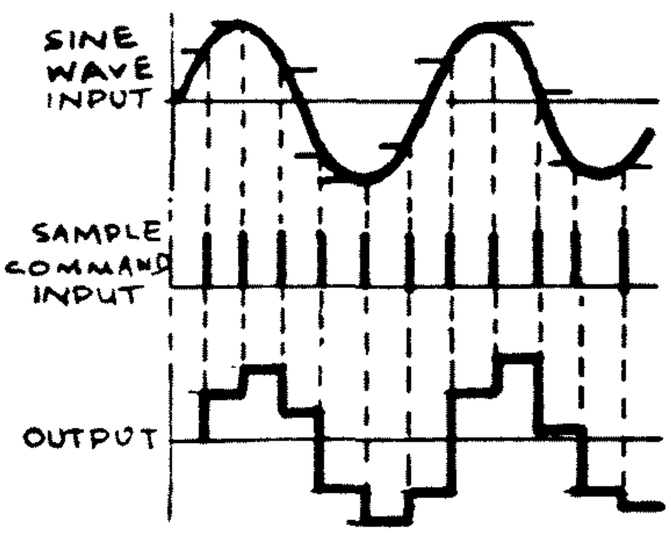
When a gate voltage is presented to the generator input, the output voltage rises, at a rate determined by the setting of the ATTACK time control, to +10V. It remains at that level until the gate voltage is removed, at which time it falls to zero at a rate determined by the setting of the RELEASE time control.

If the gate signal is removed before the output voltage has risen all the way to +10V, the output voltage will immediately begin to fall to zero at a rate determined by the setting of the RELEASE time control.



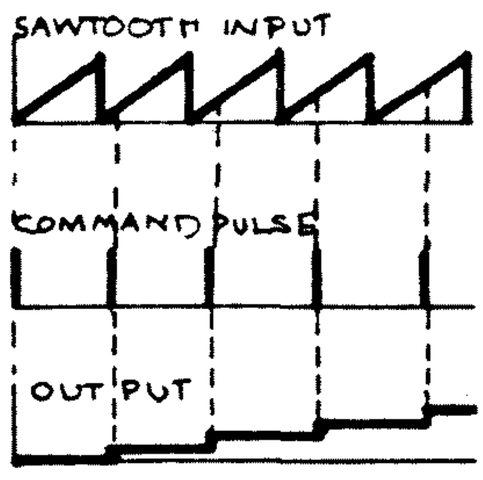
The A/R generator does not require a triggering signal. Its action is in fact very similar to that of the lag processor (see 4.53), except that the latter does not offer separately controllable rise and fall times for its output voltages.





4.32 The SAMPLE AND HOLD circuit produces stepped output voltages by sampling at intervals the instantaneous value of any signal fed to its signal input. The voltages produced in this manner are useful for control of oscillator and filter frequencies and occasionally of VCA gain. Other more exotic possibilities may occur to the experienced user.

The S/H circuit is provided, in addition to a signal input for the waveform to be sampled, and a signal output giving the result of the sampling operation, with a "sample command" input. On the 2600 this is normally connected to the INTERNAL CLOCK, and so the jack that interrupts this connection is labelled EXTERNAL CLOCK. Any square or pulse wave on the 2600, or the keyboard gate or trigger signals, are appropriate inputs to this jack.



When a pulse is applied to this input, the output signal voltage immediately assumes the same value as the input signal voltage at that instant. After the sample command pulse has passed, the output signal voltage will hold its level until another sample command pulse is presented to the circuit.

Any signal whatsoever may be sampled. The prewired connection is from the Noise Generator, but insertion of a patch cord automatically breaks this connection. The accompanying diagrams show how, when the signal being sampled is random noise, the output voltages are correspondingly unpredictable. An infinite variety of cyclical output patterns may be obtained, on the other hand, by sampling any periodic waveform. Different ratios of the sampling frequencies to the frequency of the waveform being sampled create different melodic patterns if the output voltage is being used to govern the frequency of a VCO.

The LEVEL control attenuates the input signal before it is fed to the S/H circuit. The RATE control actually belongs to the INTERNAL CLOCK; when that is disconnected from the S/H circuit, the RATE control has no effect on the operation of the S/H circuit.

The signal input is DC coupled.

The sample command input is AC coupled, and will trigger reliably from any sharp rising waveform that:

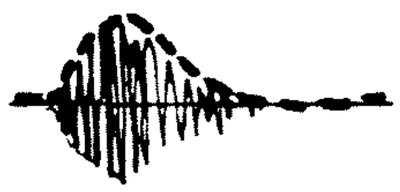
a) has a risetime of less than 10 microseconds,

b) has an amplitude of at least 5V, and

c) lasts longer than 20 microseconds.

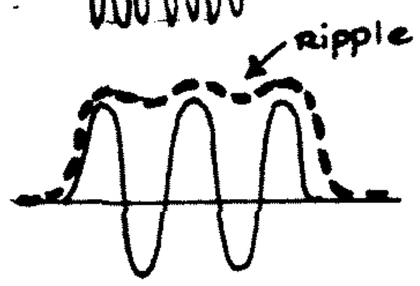
The sample command input impedance is not less than 50Kohm.
Sampling time is less than 25 microseconds.
Drift is less than 10 microvolts/second.
Output impedance is 1Kohm.

4.33 The ENVELOPE FOLLOWER generates from any audio-frequency input an output voltage directly proportional to the AVERAGE of the input signal AMPLITUDE. Its sensitivity is such that, with the input attenuator wide open, a 1V P-P square wave will produce a +10V output. The maximum output is +10V.



The risetime, or time it takes for the envelope follower to respond to any sudden change in the amplitude of the signal input, is 10 milliseconds to 50% of final value and 30 milliseconds to 90% of final value.

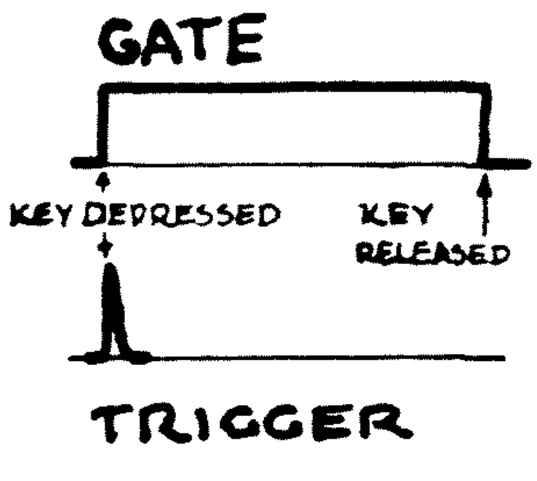
Like all similar circuitry, the envelope follower tends to "ride" on low audio frequencies as if they themselves represented changes in *amplitude; this is not critical* but has been held to a ripple of less than 1% P-P down to 100Hz and less than 10% down to 40Hz.



The primary use of the envelope follower is with external instruments. Essentially it extracts from any input a control voltage representing the amplitude-envelope of that signal: this voltage may be used to control the VCF, VCA, or any of the VCO's. It should in general be enough to remember that the envelope follower output is an envelope and can be used in the same fashion as the output from either of the envelope generators. Other uses may suggest themselves to the experienced.

The input impedance is 100Kohms and the output impedance is 1Kohm.

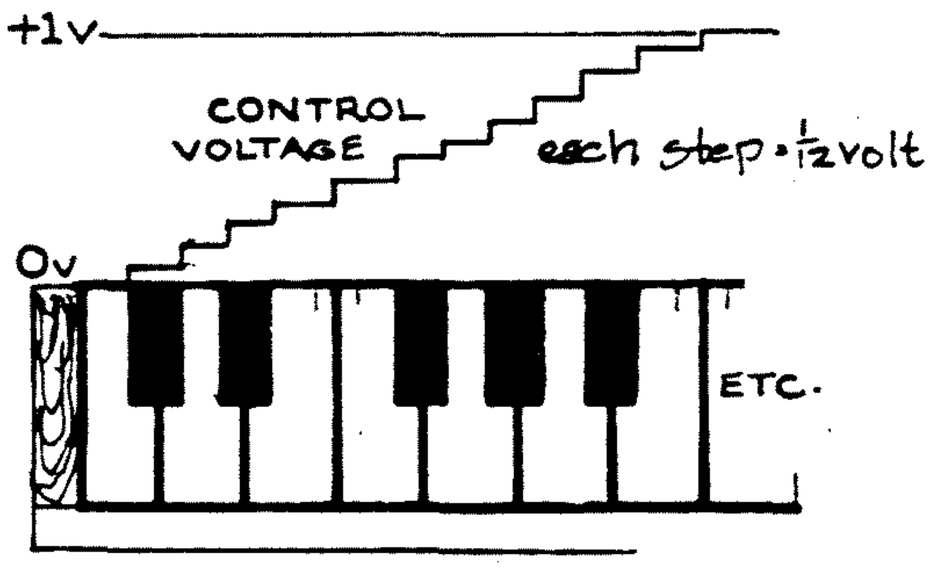
4.34 The KEYBOARD CONTROLLER 3604 has three outputs: these are a GATE signal, a TRIGGER signal, and a CONTROL VOLTAGE signal. When the keyboard is connected to the 2600 these signals are automatically routed to the functions they are most often used to control. The GATE and TRIGGER signals are fed to the ENVELOPE GENERATORS and the KEYBOARD CONTROL VOLTAGE (KBD CV) is



fed to the three VCO's and the VCF. In addition, the KBD CV becomes available at the output jack located in the extreme lower left corner of the 2600 main unit, and the GATE and TRIGGER outputs are available at two jacks located in the Envelope Generator section.

When any key is depressed, the GATE voltage immediately rises to +10V and remains at that value until the key is released again, upon which the gate voltage returns immediately to zero. If more than one key is depressed simultaneously, the gate voltage does not fall to zero until they have all been released. The function of the gate voltage, in other words, is only to indicate that at least one key is depressed.

At the instant a key is depressed, the TRIGGER voltage rises to approximately +15V and instantly returns to zero. This trigger output appears EVERY time a key is depressed, regardless of how many keys are already being held down. The function of the trigger, in other words, is to indicate the exact instant at which any key is depressed. Releasing a depressed key has no effect on the trigger output.



When any key is depressed, the KEYBOARD CONTROL VOLTAGE immediately assumes some value from OV to somewhat less than +8V depending on which key is depressed and on the settings of the INTERVAL and TUNING controls at the front of the keyboard. The output will remain at this voltage until another key is depressed, even after the first key is released: a "memory" circuit in the keyboard holds the voltage until it is changed by depressing a different key. If more than one key is depressed simultaneously, the control voltage output will be governed by the lowest, or leftmost, key, and will not be affected by depressing any key to the right of that key.

4.341 At the front of the 3604 Keyboard Controller are several controls that affect the behaviour of the keyboard control voltage. Each is activated by a switch and governed by a knob.

The TONE INTERVAL control is essentially a precision attenuator for the control voltage. When the slide switch for this control is in its RIGHT-HAND position, the output voltage passes through an attenuator that is factory-calibrated to give exactly one volt output per keyboard octave, or four volts from one end of the keyboard to the

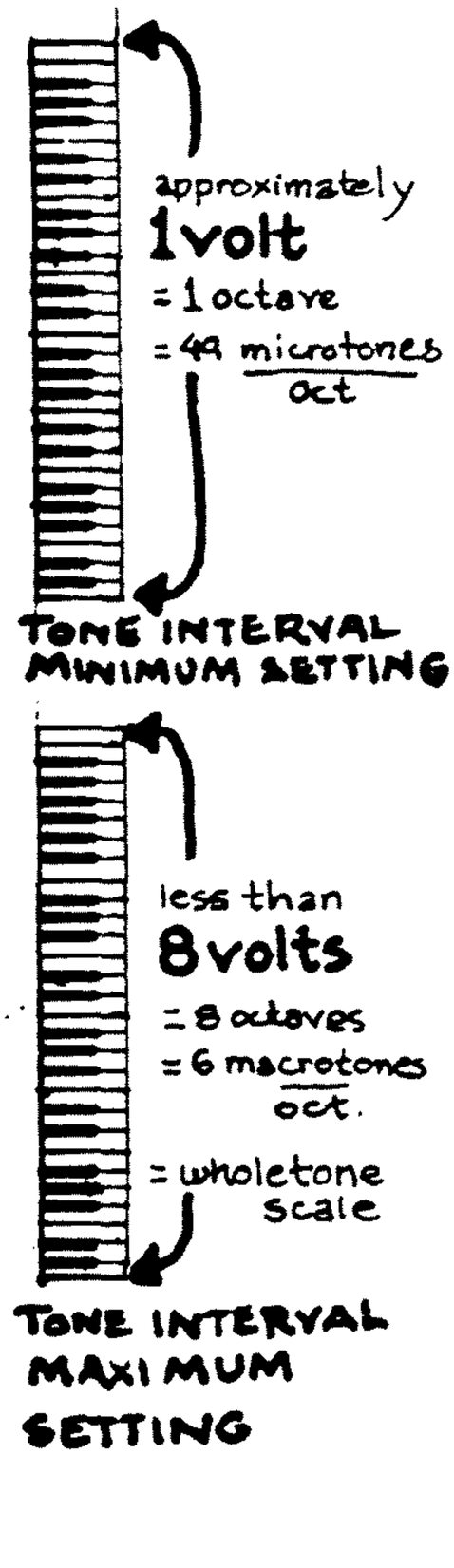
other. Thus, under these conditions, when the KBD CV controls any VCO in the 2600, the VCO will respond with one audio octave interval for every octave interval played on the keyboard; and intermediate keyboard intervals will produce the standard musical intervals of a tempered scale.

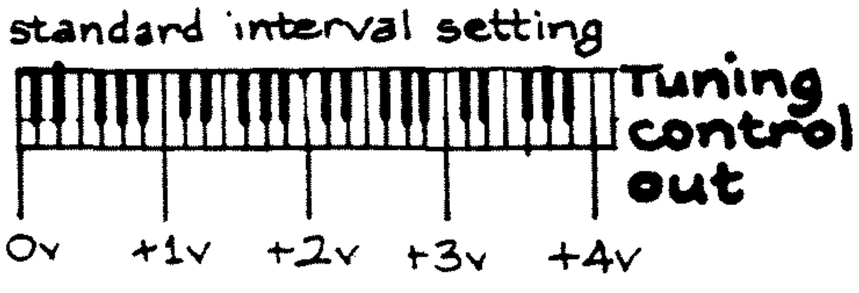
When, however, the slide switch is in its left-hand position, the KBD CV passes through a rotary attenuator that may be user-adjusted to a wide range of values. With this knob turned to its maximum counterclockwise position, the voltage differences produced from one end to the other of the keyboard is only about one volt, or (when controlling a VCO) a musical interval of about one octave. This octave is therefore divisible into as many microtones as there are keys on the 3604.

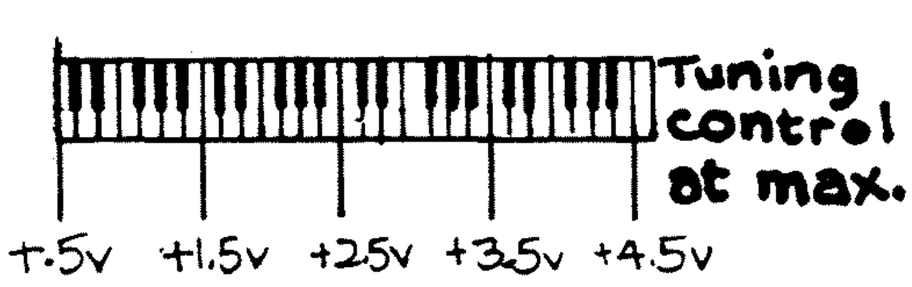
When the TONE INTERVAL knob is turned to its maximum clockwise rotation, the voltage interval from bottom to top of the keyboard is not quite eight volts, or (when controlling a VCO) a musical range of somewhat less than eight octaves.

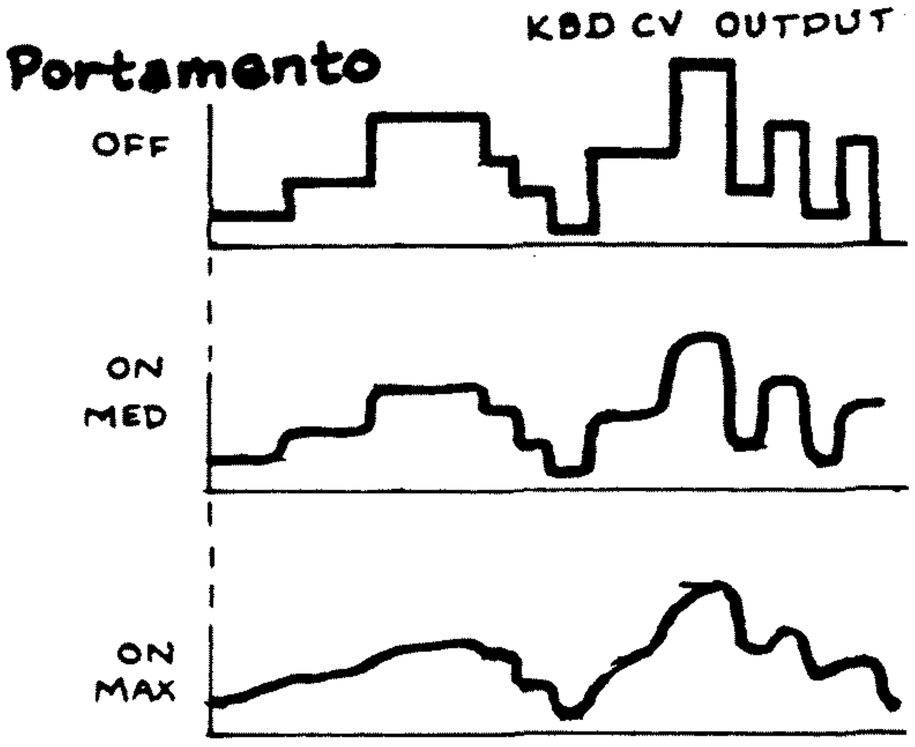
Thus the musical interval represented by the four keyboard octaves of the 3604 may be adjusted over a wide range by the TONE INTERVAL control.

Normally the voltage represented by the lowest note on the keyboard is zero. The entire keyboard output voltage may however be adjusted over a +.5V range from this reference by use of the TUNING control. When the switch for this control is in its right-hand position, the control is disconnected and has no effect on the output voltage. When the switch is in its left-hand position, the TUNING knob as it is rotated clockwise will gradually raise the KBD CV to about a half-volt higher than its corresponding value with the TUNING control switched off. By means of this control, when the 2600 is being used with other instruments it may first be brought into tune with itself and then tuned to the other instruments (provided the initial tuning is within half an octave below the intended pitch).



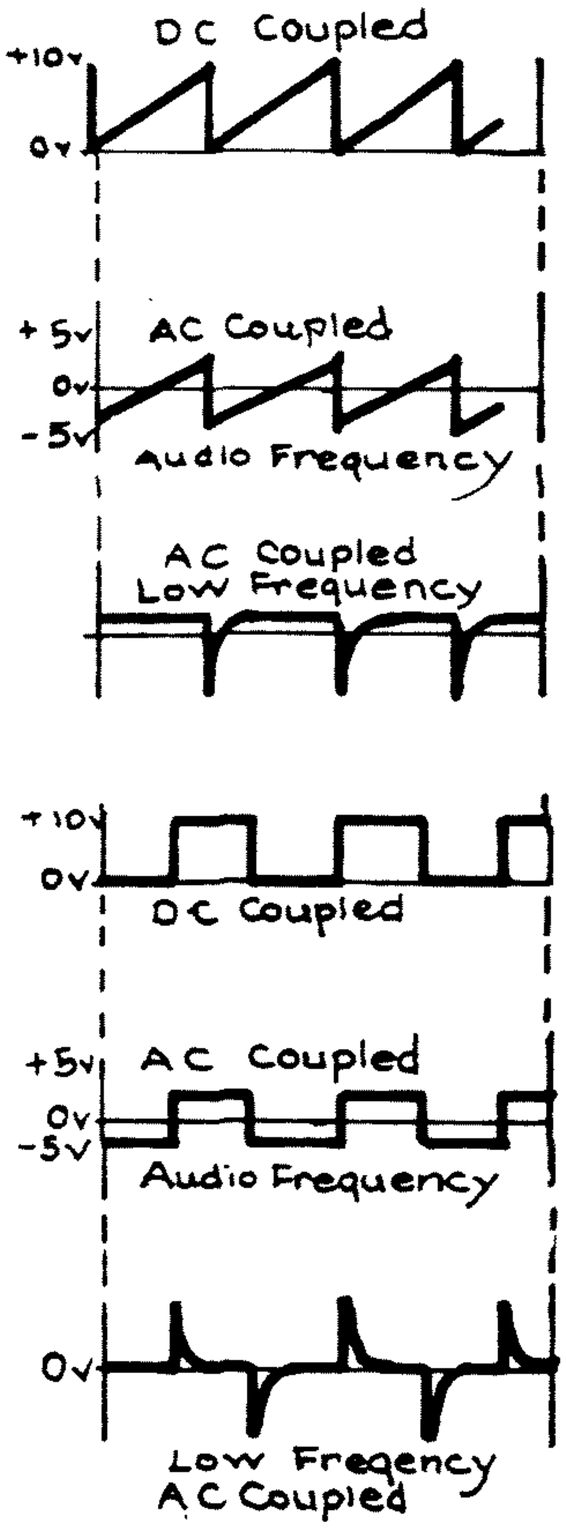






The PORTAMENTO control introduces, when it is switched on, a variable slide into the control voltage output. Electrically its action is very similar to that of the Lag Voltage Processor. Its effect is to prevent the control voltage output from responding instantly as different keys are depressed; it moves more slowly from one value to another. This in turn means that a VCO being controlled from the keyboard will not move sharply and quickly from one pitch to another but will "slide" from one to another of the pitches dictated to it by the keyboard.

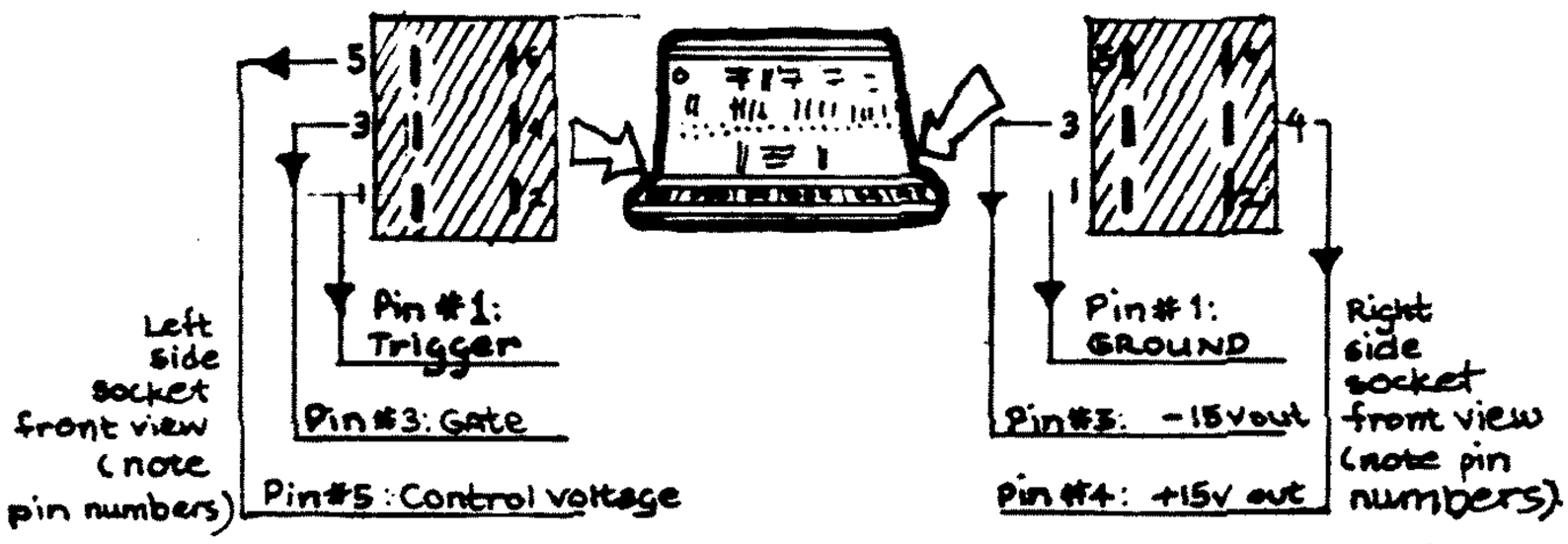
4.341 WIRING OF THE KEYBOARD SOCKETS is given in the marginal diagrams. The socket on the right side of the 2600 is the POWER connection to the keyboard and the socket on the left side of the 2600 is the INPUT from the keyboard. If for any reason it is desired to extend the short keyboard connecting cables, great care should be taken to wire the added connectors properly, and not to interchange them in use. Both the 2600 and the 3604 could possibly be damaged by incorrect wiring of the connectors.



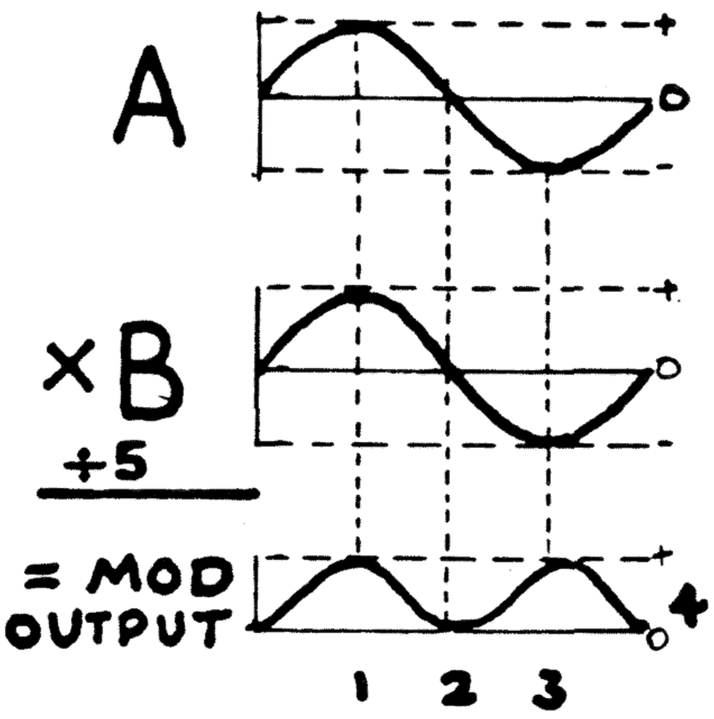
4.4 The RING MODULATOR is essentially a voltage multiplier; from two inputs A and B it produces the output function A x B/5. The kind of transformation this effects on input signals depends to a large extent on what they are and on whether the modulator is AC or DC coupled to them. The coupling is selected by the AUDIO-DC switch at the bottom of the modulator.

When the inputs are AC coupled (AUDIO position of the switch), any DC component present in them is cancelled before they are fed to the modulator. Thus a sawtooth that starts from zero and goes to +10V will instead start at −5V and move to +5V so that its overall positive and negative deviation cancels to zero. Under these conditions the modulator will generate from any two periodic signals an output signal consisting of the sum and difference frequencies that can be generated from the frequencies of the two inputs. The input frequencies themselves will be suppressed.

If both signals are audio-frequency, a large variety of harmonic and inharmonic timbres can be produced from the modulator, depend-

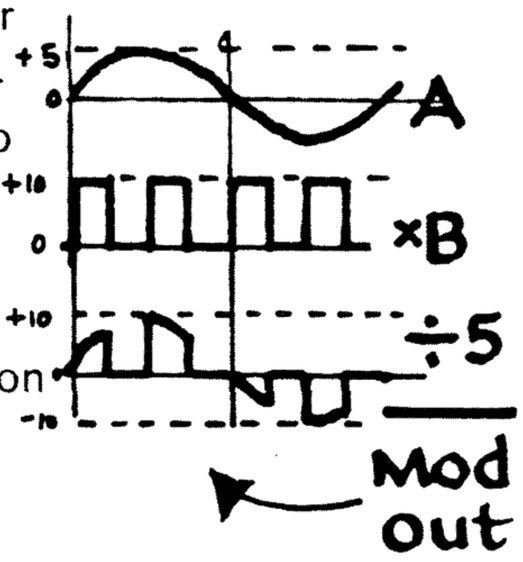


ing on the ratio of the input frequencies and on their own harmonic content. If A is a sine wave and we represent its frequency by F_a, and B is a complex waveform of frequency F_b with overtones $2F_b$, $3F_b$, $4F_b$, etc., then the output of the modulator will be a complex waveform with frequency components $F_b + F_a$, $F_b - F_a$, $2F_b \pm F_a$, $3F_b \pm F_a$, $4F_b \pm F_a$, etc. A moment's experimentation with the prewired SAW-TOOTH and SINE inputs to the modulator will demonstrate the complexity of the timbres that can be generated by this simple means.

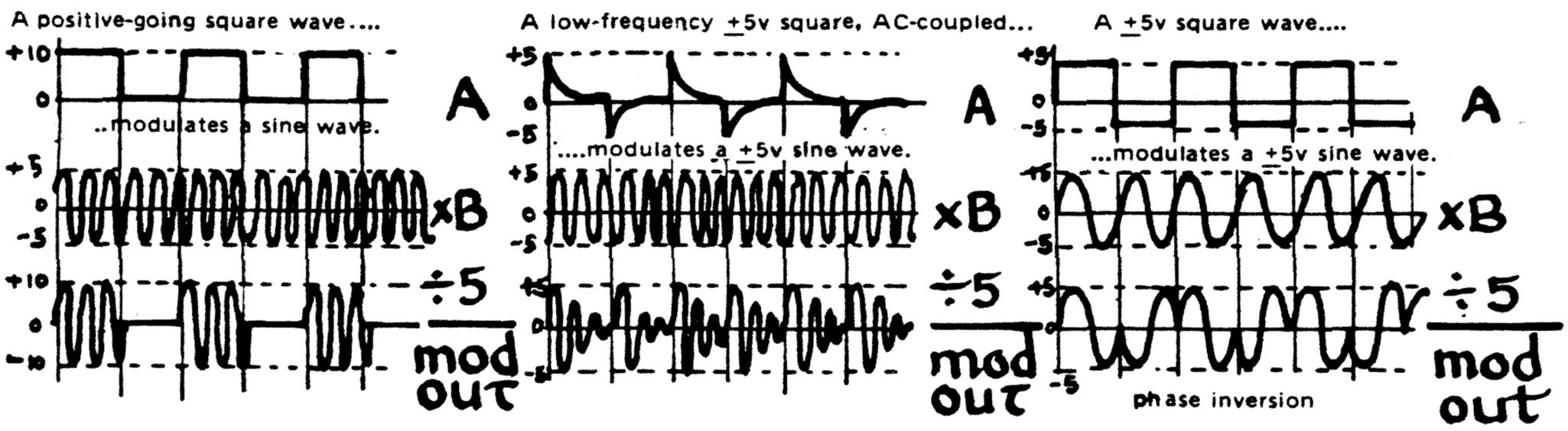


If, still with AC coupling, one input is subsonic and the other at some audio frequency, there will be an output from the modulator only when the value of the subsonic input is changing, and the output will be roughly proportional to the rate of change. If, for example, the subsonic input is a square wave, the modulator output will be a series of short, decaying tonebursts—one at each rise or fall in the input voltage.

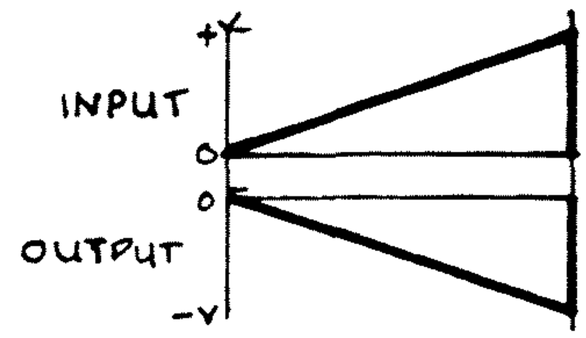
When the inputs are DC coupled, any DC component in either one of the inputs will pass into the modulator and affect the modulating process. The effect when both inputs are at audio frequency is to allow into the output waveform some of the input frequencies in addition to the sum and difference frequencies. The effect when one of the inputs is subsonic is that the modulator operates as a voltage-controlled amplifier: the output amplitude will be in direct proportion to the instantaneous amplitude of the low-frequency input and will vary as its absolute value varies. Also, the output phase will reverse when the low-frequency input voltage changes from positive to negative or vice versa.

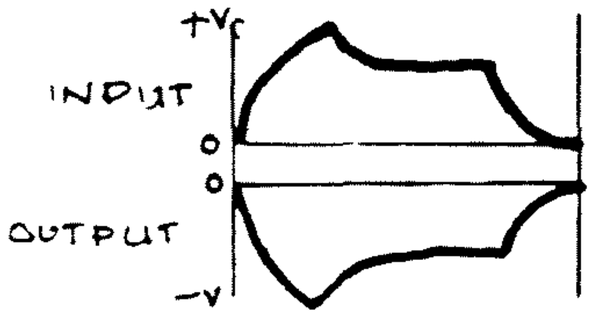


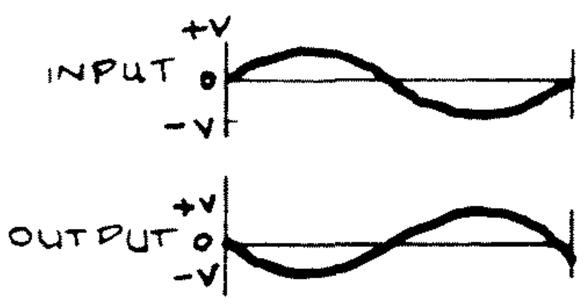
The input impedance is greater than 15Kohms and the output impedance is greater than 1Kohm. The signal-to-noise ratio is better than 60Db referred to a 10V input, and carrier rejection is greater than 80Db for the left input and greater than 60Db for the right input. AC-coupling time constants are 235 msec. for the left input and 90 msec. for the right input.



VOLTAGE PROCESSOR (INVERTER)







4.5 The VOLTAGE PROCESSOR section of the 2600 contains three separate processors. Two are mixing and inverting processors and the third produces a variable lag.

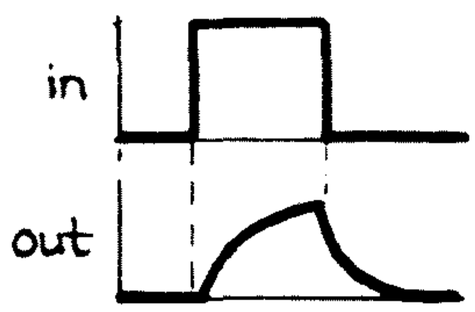
4.51 The first processor accepts up to four inputs in the range from DC through 20KHz and produces an output voltage of opposite polarity from that represented by the instantaneous sum of the input voltages. Thus for example the keyboard voltage, which normally goes from 0V to +4V, if run through the inverting processor would go from 0V to −4V and would therefore drive the frequency of an oscillator down from left to right on the keyboard. Positive-going envelopes become negative-going envelopes, etc.

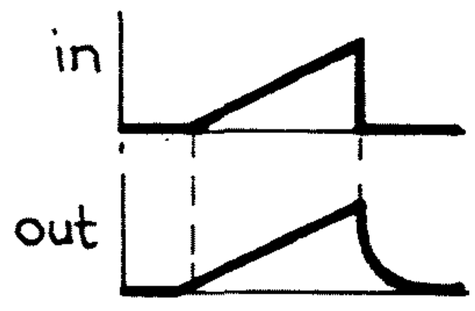
This inversion, applied to audio signals, puts them exactly 180° out of phase with their uninverted form. Such phase inversion has absolutely no audible effect unless an inverted signal is in some form mixed with its uninverted counterpart, in which case the signals mixed will partially or entirely vanish.

Two of the inputs to the first inverting processor are attenuator governed and two are not. The inverter has one output.

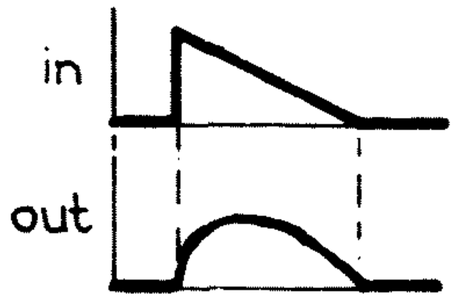
4.52 The second inverting processor is exactly the same as the first except that it has only two inputs, one attenuator-governed and one unattenuated.

LAG VOLTAGE PROCESSOR





4.53 The LAG PROCESSOR acts only on sudden changes in the input voltage presented to it. It slows them down by an amount controlled with the slider in the signal path. With this control set at maximum, the time required for the output voltage to reach a value suddenly assumed by the input voltage is at most about .5 sec. With the control at minimum the amount of time required is at most about 5 ten-thousadnths ofa second. This minimum lag has no audible effect when the signal being processed is a control signal of low frequency, but it is there nonetheless: audio-frequency signals will be filtered sharply above 1600Hz by even the minimum lag time of the processor. The time constants strictly defined depend somewhat on the output impedance of the signal source; for a 1Kohm source impedance they are .1ms minimum and .1sec maximum.



Because of its effect on audio signals the lag processor can double as a low-pass filter with maximum F_C of 1600Hz. Minimum F_C is subsonic at 1.6Hz. When it is used in this fashion, the cutoff slope is –6Db/Octave above F_C.

The input impedance of the voltage inverters is 50Kohms minimum and of the lag processor 1Kohm minimum above 1.6KHz.

4.6 The PERIPHERAL EQUIPMENT includes the MICROPHONE PREAMP, the ELECTRONIC SWITCH, the INTERNAL CLOCK, the MIXER and PAN POT, and the REVERBERATION UNIT. All of these are extremely simple functions with few or no controls except input or output attenuators.

4.61 The MICROPHONE PREAMP accepts any low-to-medium impedance input. A rotary attenuator controls the level of the input, and a three-position slide switch selects one of three operating ranges; 20, 40, or 60Db gain. The preamp is useful not only with microphone signals, but also with direct inputs from electric instruments and for some purposes even with signals internally generated by the 2600 itself. Working with a signal of unknown strength, it is best to begin with the lowest gain in finding the best operating range for the preamp.

4.62 The ELECTRONIC SWITCH is bidirectional. That is, it will either switch a single input alternately to its two outputs, or it will accept two inputs and switch them alternately to one output. The switch is useful both for audio and control signals.

4.621 INTERNAL CLOCK: this is simply a square-wave oscillator serving a particular purpose. Its output is permanently connected to the electronic switch; it is prewired to the sample command input of the Sample and Hold circuit and to the Envelope Generator's external gate input. The frequency of the INTERNAL CLOCK is governed by the RATE control and covers a range of approximately .2Hz to 100Hz. The output from the clock is available independently at the INT CLOCK OUT jack.

4.63 The general purpose MIXER accepts two inputs through attenuators and feeds their sum to the panpot. Note the two jacks just above the attenuators: insertion of a patch cord to one of these jacks disconnects the attenuator output from the mixer input. These are output jacks and make it possible to use the mixer input attenuators

as floating signal-or control-voltage attenuators wherever they might be needed. The mixer is good down to DC voltages and can be used to mix control signals as well as audio signals.

4.631 The PANNING POTENTIOMETER (Panpot) has one input, normally from the mixer. When the control slider is centered, the input signal is sent in equal proportions to the left and right outputs. As the slider is moved to the left or right, the signal level is decreased on the opposite channel until at either extreme position, it disappears entirely. The audible effect is that by means of the panpot it is possible to "position" a signal anywhere in the stereo output of the 2600.

4.64 The REVERBERATION UNIT has one input, normally prewired from the mixer output. The signal presented to this input is equalized and sent to a pair of reverb springs. The spring output, amplified, is sent to two attenuators; the signal to one of them is phase-inverted to create a strong sense of spaciousness in the reverberated output.

From the left-channel attenuator the signal passes directly to the left-channel output of the 2600; the right-channel attenuator output is fed through the REVERB OUTPUT jack to the right-channel output of the 2600. Inserting a patch cord into the REV OUT jack interrupts this connection and makes a reverberated signal available for feeding to other functions in the 2600.

4.65 Outputs from the PANPOT, REVERB UNIT, and LEFT and RIGHT INPUTS (near the upper right corner of the 2600) are combined at summing points for each channel. The LEFT CHANNEL signal is available at the LEFT OUTPUT jack and is internally fed to a 1-watt power amplifier which drives the LEFT CHANNEL MONITOR SPEAKER in the 2600. The RIGHT CHANNEL signal is available at the RIGHT OUTPUT jack and is fed internally to a 1-watt power amplifier which drives the RIGHT CHANNEL MONITOR SPEAKER. The volume from each speaker is governed by an attenuator immediately next to it.

These attenuators also govern the signal fed to the HEADPHONE jack at the lower right corner of the 2600, just under the power switch. Insertion of a standard stereo headphone jack automatically disconnects the monitor speakers. Headphone impedance should be 8 ohms.

4.66 The MULTIPLE OUTLET at the left side of the control panel may be used to distribute a single output to as many as three inputs. It is like a series of "Y-connectors".

5.0 PUTTING IT ALL TOGETHER: is your job. We can't do it for you. But what we have done in this section is to suggest, first of all, a plan of organization so that as you really get into your instrument you don't go out of your skull trying to remember it all. This plan of organization is inherent in the way we have structured the chapter. Second, we give you the results of our own work with the 2600 synthesizer: suggestions for special-purpose patches, effects, and so on. Third and probably most important, we encourage you to experiment on your own, systematically at first and then with more and more freedom as your familiarity with the synthesizer grows.

5.01 Your investigations should always be governed by two principles, in a sense opposites of each other. One is that no damage can be done to the 2600 by any internal connections you can possibly make. So a voltage is a voltage is a voltage; except in a formal, functional sense, there is no difference between signal and control voltages. There are no "forbidden" connections on the 2600.

5.02 The other principle is that, on the other hand, not all connections are useful. And here we can do no more than to suggest the principle: we cannot undertake to define for you what patches are useful and what patches are not. What is useful in the studio is sometimes too clumsy for live performance, or too time-consuming to set up, or might involve too critical a tuning preparation. Or what is useful in live performance might be too crude or simplistic for studio use. And patches designed for instructional purposes, in a classroom, might have no musical value at all.

5.03 A few words about our procedure in this chapter. We begin from the classic Oscillator-Filter-Amplifier patch without voltage control, and build onto it until we have arrived at the classic live-performance patch 5.14. Each significant addition to the patch is illustrated with at least a block diagram; for the first few we also give a diagram of the 2600 face panel, showing the patch connections.

5.031 NOTE THAT since all the connections involved in what we have called the "classic" patch are prewired into the 2600, activating them is simply a matter of raising, or opening, a few attenuators. This remains true for many of the more complicated or unusual patches that we deal with later on; accordingly, on our block diagrams we indicate with the abbreviation "p. cord" those connections that are not part of the 2600 prewired patch and must be made by the user with patch cords.

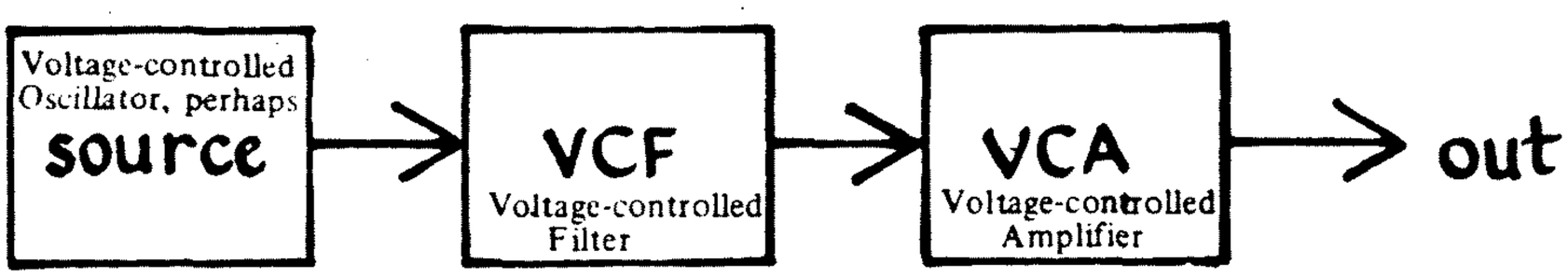
5.0311 NOTE TOO that because of the prewired patch connections, it is important that all input attenuators not involved in a particular patch must be COMPLETELY CLOSED, i.e. all the way down. Failure to observe this precaution will result in a great deal of confusion.

5.032 The abbreviations used in the block diagrams are derived from the initials of the 2600 functions. Thus "EF" is the envelope follower, "MP" the Microphone Preamp, and so on. Since there are three separate voltage processors we refer to them by their function; "VPinv" or VPlag". To simplify the diagrams we allow control signals to leave either from the right side or the top. Ring Modulator inputs are drawn according to the way the modulator is functioning in each patch.

In general signal paths are from left to right and control paths from bottom to top.

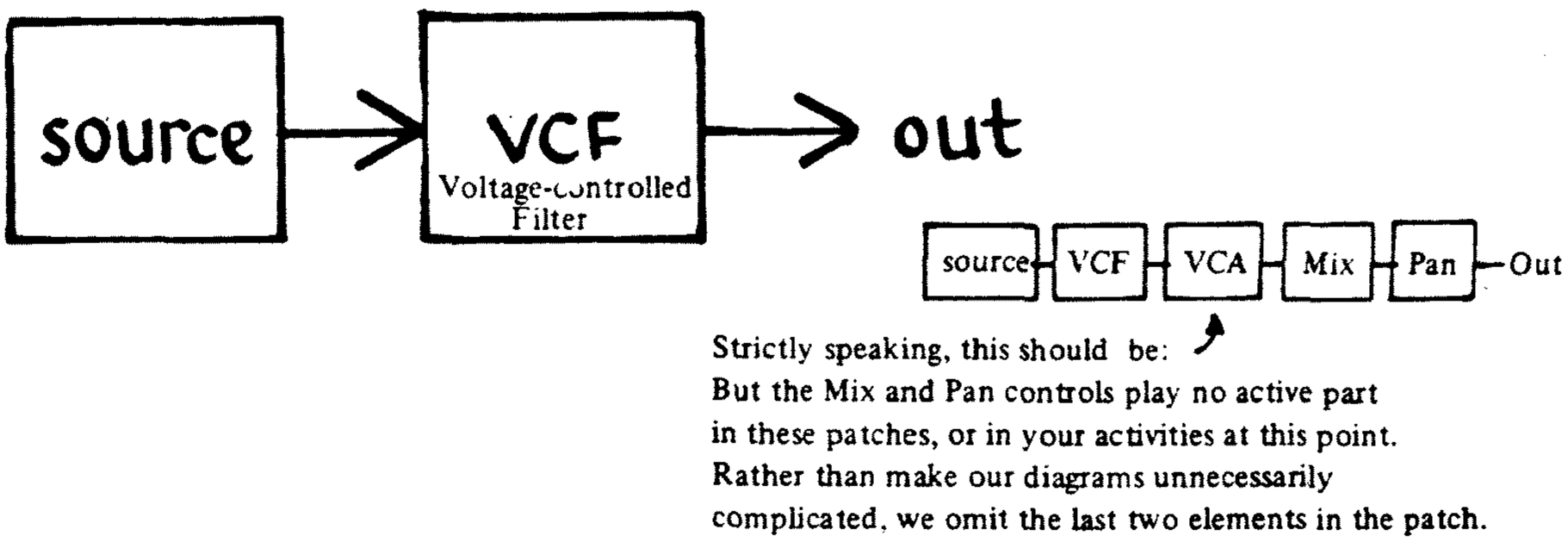
5.04 This chapter is designed to be user expandable with additional patch diagrams, block diagrams, performance notes, etc. To make this easier, we have, as far as possible, restricted the later portions of the text to two numbered sections on each page. Between any two pages, therefore, another page can be inserted and numbered in logical subordination to the other pages. This system is indefinitely expandable. From time to time, as 2600 users feed back to us the results of their experience, we will send out patch and use notes to be inserted in their proper places in this chapter.

5.1 THE CLASSIC PATCH AND ITS EXTENSIONS.



. . . *in which an oscillator or noise output at some audio frequency* is filtered and amplified.

This is the way it was done in the days before voltage control, and still forms the basis for many patches you will find useful. As the block diagram indicates, we are talking here only about the path of an audio signal through a filter and an amplifier. Since the only point of the amplifier's presence in the signal path is to provide amplitude control over the signal, it could perfectly well be replaced so long as we are concerned only with manual control by a simple attenuator. If that were done the patch would essentially reduce to



which is about as basic as you can get.

(This section is going to be very wordy while you're getting your bearings. By the time you reach 5.14 or so they'll be moving a little faster.)

Use this basic patch to acquaint yourself with the behavior of the oscillators including their frequency ranges, tonal character of their audio waveforms, and so on; with the action of the low-pass filter, on signals fed into it, and with the dispensability of the amplifier when it is subject only to manual control. Let's do this one thing at a time.

Begin with the following settings (and return to them whenever later in this chapter, we say something like "start from the beginning" or words to that effect. If you need to know "where to start", this is it!):

a) All input attenuators and output level controls at their minimum, or off, position. This is always down for vertical sliders and left for horizontal sliders.
b) All VCO control sliders, including the pulse-width controls on 2 and 3 centered; range switches to audio position.
c) VCF initial frequency and fine tune sliders at maximum, i.e., extreme right; resonance slider at minimum, i.e. extreme left.
d) all envelope-generator control sliders at minimum; gate selector switch to keyboard—upper—position.
e) VCA initial gain control to maximum—right—position.
f) panpot centered, reverb minimum.
g) noise generator controls centered.
h) on the keyboard, all three switches at the front should be in their right-hand position.
i) anything we haven't mentioned either is covered implicitly in a—g or doesn't matter.
j) the best course to take with the speaker volume controls is to leave them about three-quarters of the way open and then change the volume whenever you feel like it.

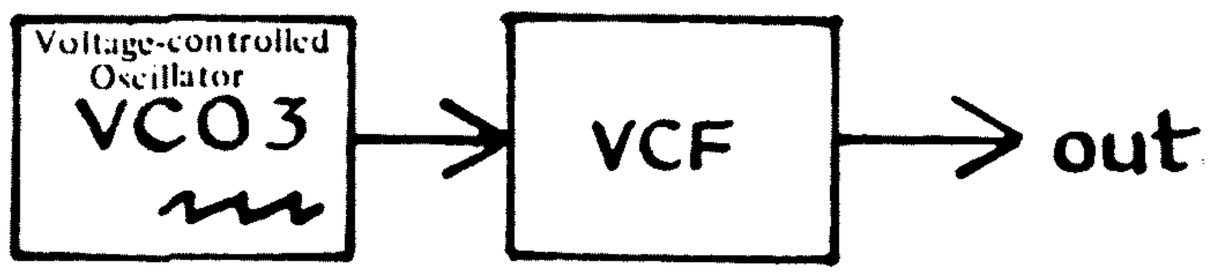
NOW

5.101 Open fully the Mixer input from the VCF and the VCF input from VCO-3 (after this we'll refer to these as simply "VCF-Mixer" and "VCO3-VCF" inputs respectively, and use the same shorthand for other prewired connections); you should hear a medium-low, humming tone from the speakers.

(If you don't, ask yourself as systematic a set of questions as you can about the signal path you just set up and about what operating characteristics you have set for each function in the path. In the course of answering your questions you will discover that you've overlooked something. For the present case we suggest the following. 1) Is VCO3 in its audio range? 2) Have you by accident pressed one of the upper keyboard notes and driven VCO3 out of audible range? Remember that the keyboard will hold for an indefinite period of time the last voltage keyed from it; develop the habit of flicking the lowest key before setting any VCO initial frequency controls or attempting to tune two or more VCO's to any particular pitch or pitch interval. 3) Is the VCF initial frequency control set to maximum? Is the resonance control set to minimum? 4) Was it really the input from VCO3 to the filter, or some other one, that you opened? Was it really the filter input to the mixer that you opened, or some other one? 5) Is the speaker volume high enough? 6) Is the power on? Is the unit plugged in? If such a check procedure doesn't reveal anything wrong, and if another one still doesn't show any errors in setup, and there still isn't any sound, your 2600 is in need of service. Call your dealer immediately. If, on the other hand, you discover and rectify an error in the patch or in your control settings, you will indeed hear the sound of VCO3 pouring from the loudspeakers, and you're on your way. We resume our narrative.)

This sound is the VCO3 sawtooth waveform, essentially unmodified, at a frequency somewhere below 1KHz.

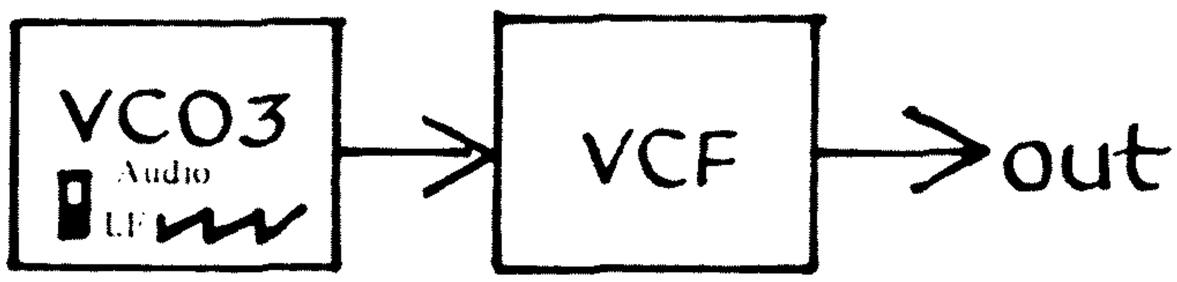
Experiment on it: use the VCO3 frequency sliders to raise and lower the pitch. You will discover that the initial frequency slider can raise and lower the pitch to above and below audibility without assistance from the fine tune slider. The fine tune slider can only change the pitch by about one octave, certainly not by more than two. The keyboard control voltage "plays" the oscillator as though it were a flute or penny-whistle—play a scale on the keyboard and the keyboard control voltage will play a scale on the VCO. But using a control voltage doesn't disconnect the manual oscillator control, not at all: play a keyboard scale with one hand while with the other you move the initial oscillator frequency control.



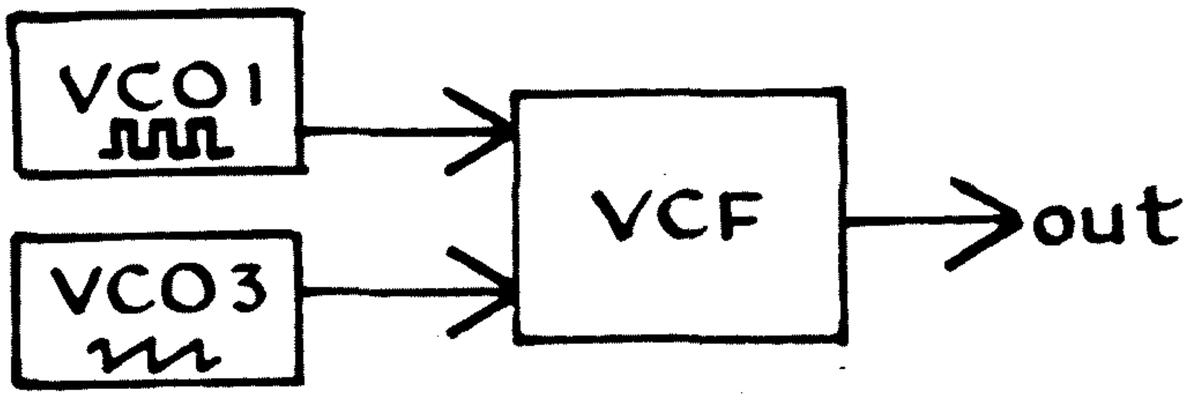
Experiment some more: switch the VCO3 range switch to LF and note what happens. The first time around you may even miss the periodic "click" as the VCO3 output voltage, now at some subsonic frequency, rises slowly to its maximum value and then suddenly drops again to zero. (It's the sudden change once in each cycle that you hear as a click; its mechanical effect on the loudspeaker is exactly the same as if you were to tap the cone with a pencil. The slow ascending portion of the VCO output voltage has no audible effect on the loudspeaker.) Use the frequency control sliders again and note that the only apparent difference they make is in the length of time between clicks; only at the extreme high end of their range do the clicks get close enough together to produce a sense of pitch.

At the low end of its range, the VCO should be taking about thirty seconds to complete one cycle. Listening to the VCO3 sawtooth, this means about one click in thirty seconds.

This completes your preliminary examination of the VCO3 sawtooth output. Close the VCO3-filter input.



5.102 Open the VCO1-Filter input. The prewired input from VCO1 is a square wave and sounds considerably different from the sawtooth. Briefly familiarize yourself with its range of control as you did with VCO3. Note particularly that in the extreme low-frequency range, the square wave produces <u>two</u> clicks for every one produced by the sawtooth. This is because the square wave has two sharp voltage changes in each cycle, each one of which produces a click.



By opening both the VCO1 and VCO3 Filter inputs about halfway, you can listen to both oscillators simultaneously. Do that; if you like you can play with the resulting click patterns for a while before getting on to more serious business, namely

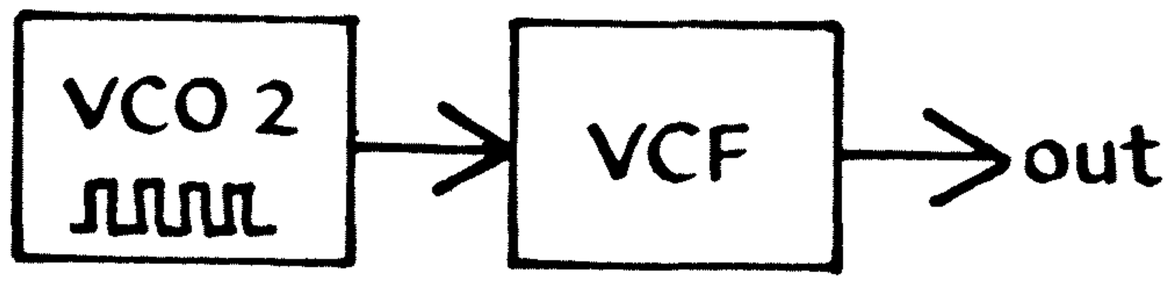
5.1021 TUNING TWO OSCILLATORS TO UNISON. Switch both VCO's back to audio range and set one of them to a comfortable pitch. Now tune the other one to unison with it by "zero-beating" the summed output you are listening to. The procedure for zero-beating is to 1) first bring the two oscillators as close together in pitch as you can by using the initial frequency sliders, 2) then use the fine tune slider to arrive at a unison. As the oscillator frequencies approach unison you will hear a pronounced "beating" effect as the two waveforms alternately reinforce and cancel each other. This beat will occur at a frequency equal to the difference between the frequencies of the two oscillator outputs; thus if VCO1 is putting out a frequency of 1000Hz and VCO3 an output of 1001Hz, the summed output of the two will grow alternately louder and softer at the rate of once each second, or 1Hz. Summed frequencies of 1000 and 1005Hz will beat five times per second, or 5Hz. And so on. Refer to the diagrams in section 6.

Tuning to unison is therefore a simple matter of slowing down the beat frequency until it reaches an apparent zero. It's not difficult at all to reach beat frequencies of less than 1Hz; the process becomes more difficult as the beat approaches zero. By the time the beat is taking longer than five seconds per cycle you'll have to brace both thumbs against the 2600 panel and rock the fine tune slider along by the smallest possible amounts until the beat changes have ceased altogether.

The same zero-beat method is useful for tuning to fifths and fourths, though the beat becomes less pronounced as the intervals become less perfectly consonant.

Note that the keyboard control voltage, since it enters each VCO unattenuated, plays the same interval on each oscillator. Two VCO's in unison at the bottom of the keyboard should still be in unison when driven four octaves up by the +4V voltage keyed from the highest note on the keyboard.

We return again to the main course of experiments.

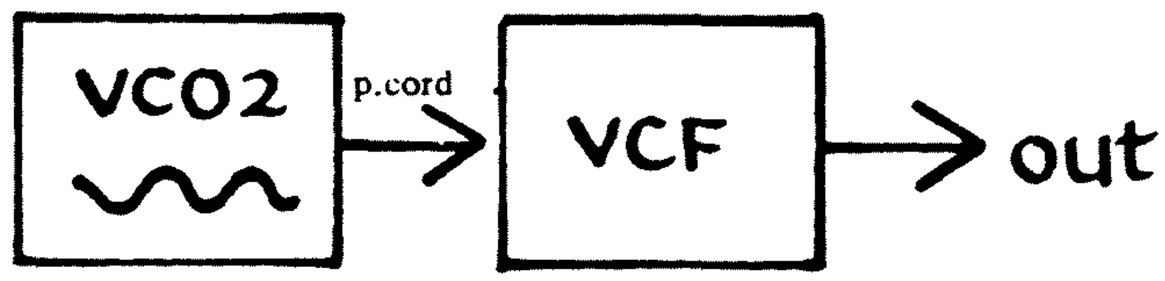


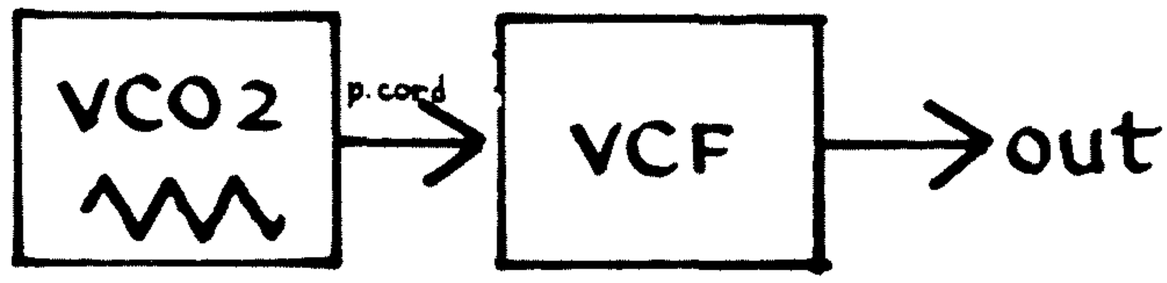
5.103 Close the VCO1-Filter input and open the VCO2-Filter input. Tune VCO2 to unison with VCO3, just for practice. Then close the VCO3-Filter input and listen to the pulse-wave output of VCO2 all by itself. This has some interesting peculiarities.

When we call this a "pulse" wave we are saying no more than that it consists of a train of pulses; but we aren't saying how long the pulses themselves last. Thus the manual control for pulse width on VCO2 adjusts the length of the pulses in the wavetrain, from a minimum length of 10% of the distance between pulses (= period of the wave) to a maximum length of 90% of the distance between pulses. To hear the difference this makes, switch VCO2 to low-frequency operation and move all three control sliders to their extreme left. The clicks you hear will be in pairs. The first click in each pair is produced by the leading, or rising, edge of a pulse, and the second click by the trailing, or falling, edge of the same pulse. Now moving the pulse-width control slider slowly to the right will increase the pulse-width and so increase the time between the first and last clicks in each pair. At the 50% panel marking, the clicks will be equally spaced; the "on" and "off" time of the pulse wave is balanced, and in fact the pulse wave is now square. Each pulse takes exactly one half cycle.

Verify that this is in fact a square wave by switching VCO2 back to audio range and tuning it to VCO1; the two should sound identical or almost identical.

While VCO2 is still in its audio range, move the pulse-width control slider slowly through its range of travel again, and note the changing tone-color of the oscillator output you are listening to. Every change in the width of the pulse changes the harmonic content of the output waveform.



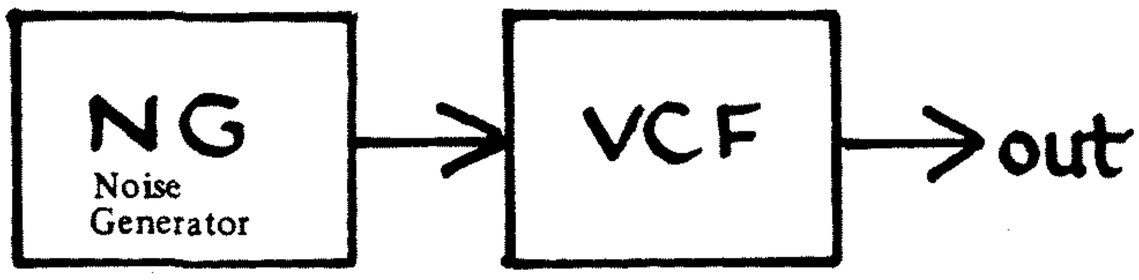


5.105 There is no reason why you shouldn't, at this point, listen to the remaining two output waveforms available from the 2600 oscillators. Take a patch cord and plug one end of it into the VCO2 sine output; plug the other end into the VCO2-Filter input (or any other, for that matter—no law says that VCO2 signals have to go to any particular Filter input). The sine wave produces a very pure, smooth tone, devoid of upper harmonics. It is a theoretical building-block from which all other waveforms can be constructed and into which they can be analyzed mathematically. To some extent these theoretical considerations are realizable in practice; some electric organs build up, in a much simplified way, their complex tones from sine waves, approximately representing the first eight harmonics or so, and the 2600's VCF is capable in a sense of resolving a complex tone into its sinusoidal elements.

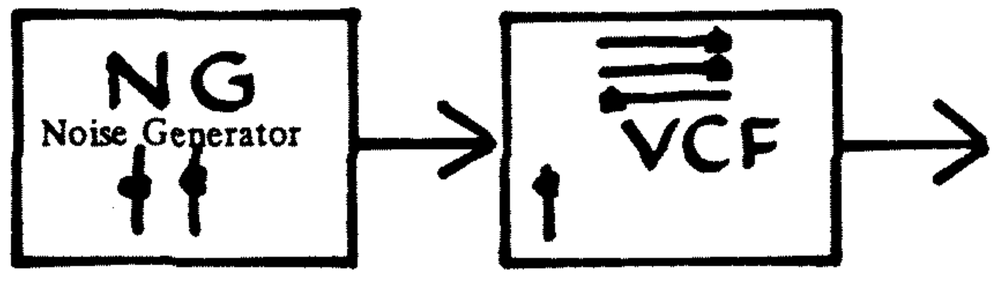
Note that because the sine wave has no sharp rises or falls in its waveform, a subsonic sine produces no clicks at all through a loudspeaker. Verify this with VCO2 in its low-frequency range. Removing the patch cord from the VCO2 sine output and plugging it into the same oscillator's triangle output will give you the sound of the triangle; as you might expect from its similarity to the shape of the sine wave, it sounds very similar but has some upper-harmonic content. Audibly it appears intermediate in brightness between the sine and sawtooth waves; tune VCO3 to VCO2 and listen to the difference between the sawtooth from VCO3 and the triangle from VCO2.

When you are familiar with all five of the oscillator output waveforms, close all the filter inputs, and remove the patchcord you were using.

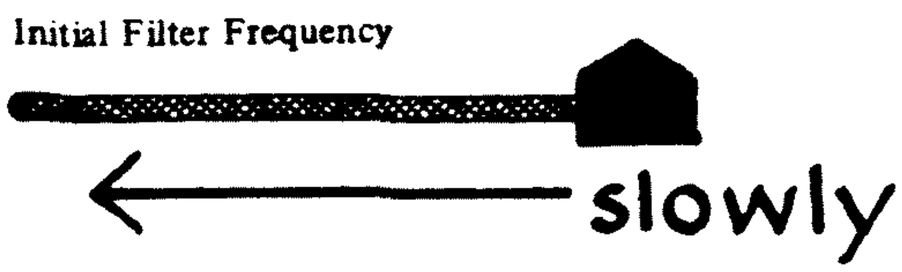
Now open the input to the filter from the



5.106 Noise Generator; this is the last of the five Filter inputs. The Noise Generator is the fourth signal generator in the 2600, and the only one that generates non-periodic output voltages. Familiarize yourself with its controls; the level control has a perfectly obvious function, and so, audibly, does the color control at its left. We have provided detailed information on the noise generator in section 4 and will not repeat it here. Our main reason for introducing it here is—aside from simply physically getting you acquainted with its two control sliders—simply to provide a logical sound source through which to introduce you to the operation and control of



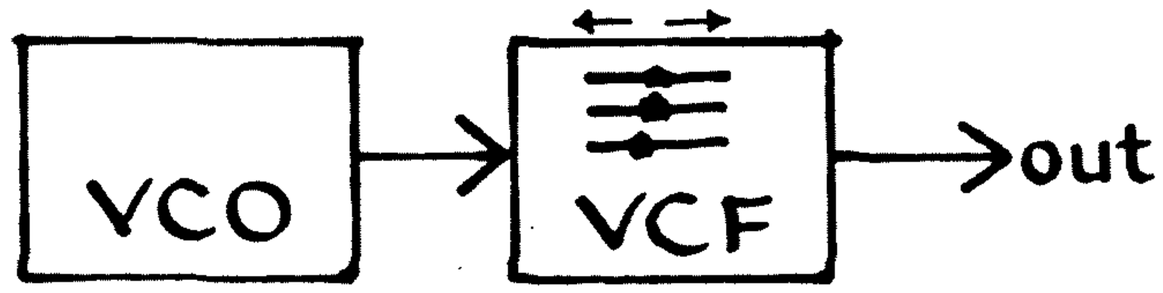
5.107 The Voltage Controlled Filter. Begin your investigation with the NG-Filter input perhaps not quite wide open—nine-tenths of maximum—the noise generator level control about two-thirds up, and the color control set for pink noise. Filter resonance at minimum and initial frequency at maximum.



Slowly lower the initial filter frequency slider from right to left. You are lowering the filter's cutoff frequency (F_c), and since the filter attenuates considerably all frequencies higher than the cutoff frequency, the noise signal you are feeding into the filter becomes, at the filter output, devoid of more and more high-frequency energy. If you refer to the first filter response graph in section 4.12 and imagine the rolloff point (indicated by the dotted vertical line labelled "F_c") moving from extreme right on the graph to extreme left, you will have a good mental picture of what you've just done to the noise input. Note that since the lowest F_c the filter is capable of is somewhat below the lowest frequency that can produce an audible pitch, the extreme left position of the frequency control slider allows no "sound" to pass at all. (*Possibly, from the noise signal, you may hear the faintest of low* rumbling noises, like distant thunder. If you set the color control on the noise generator to its lowest position you will amost certainly hear this rumbling, since you will have increased the low-frequency content in the noise signal by a considerable amount.)

Return the initial filter frequency to approximately a center position and slowly move the resonance control slider from its extreme left position towards center. Listen to the filter output fall off below F_c until almost all the noise is centered around one more or less sharply defined pitch. This should occur somewhere just to left of center. As the resonance control is moved still further to the right, the noise will disappear entirely, and the filter output will consist only of a pure sine tone. It will produce this tone even with all the inputs closed: it is oscillating, generating a sine wave independently. In this state it can function as a signal source like the VCO's.

Close the NG-Filter input and verify that the filter continues to originate a sine tone. The frequency of this output is identical to filter F_c.



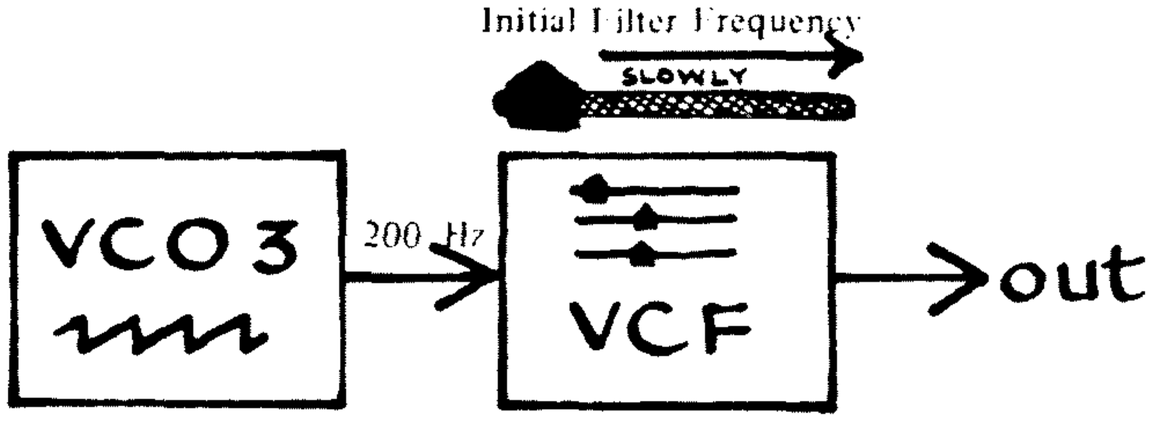
Still with all inputs closed, lower the resonance control to just below the point where the filter ceases to oscillate. At this point the filter delivers the sharpest "peaked" output it is capable of; open the *NG-Filter input again and use the keyboard this time to play up and* down on the filter F_c. For this experiment use pink noise.

5.1071 The noise signal has energy at all frequencies; it can be thought of as the sum of an infinite number of sine waves at an infinite number of frequencies. The filter, set for peaked response and swept across its frequency range, simply "picks out" and selectively emphasizes these components of the noise signal.

The periodic waveforms from the VCO's on the other hand, distribute their energy across a definite "spectrum" of frequencies. Close the NG-Filter input and open the VCO3-Filter input, with F_c at maximum and resonance still set for peaked response from the filter. Tune VCO3 down to about 200Hz.

Now flip the F_c control to minimum and start moving the control up slowly from the left. When the filter F_c reaches 200Hz you will hear a fairly loud sine tone from the speakers; this is the fundamental frequency component of the sawtooth waveform you are feeding into the filter. *Just past this point the filter output drops to almost* nothing again, until F_c reaches 400Hz, exactly one octave above the fundamental frequency of the sawtooth. Again the filter output will increase sharply and then fall off as the F_c reaches and crosses the frequency of the sawtooth wave 2nd harmonic component. The next tone will appear at 600Hz, an octave and fifth above the fundamental, and the next at 800Hz, two octaves up. This is the 4th harmonic. The 4th, 5th and 6th harmonics outline a major triad: first, third, and fifth respectively. The 7th harmonic is a minor seventh up from the 4th, and the 8th is three octaves above the fundamental. The 8th through the 16th harmonics are not quite a major scale; several of the notes in this series are sharp or flat. And so on.

(You will notice, while you are doing this, that the filter does not completely cut off frequencies below its peaking frequency. Remember that it is after all a low-pass filter, and increasing its resonance so that it peaks at F_c may attenuate lower frequencies considerably but will not cut them off. If it did it would be a bandpass filter.)

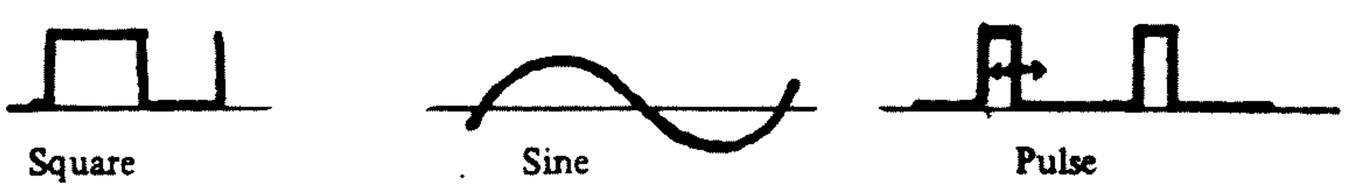


When you have reached the extreme right position of the filter initial frequency control, the peak in the filter's output will have passed pretty much beyond audibility, and the audible output will be simply a much attenuated version of the VCO3 sawtooth input. Lowering the filter resonance to minimum will restore this to its original level. It will also, of course, remove the peak from the filter output, as you will discover if, having lowered the resonance control to minimum, you sweep the initial filter frequency back from right to left. The upper harmonics will drop out but without being individually emphasized by any peak in the filter response. It becomes in fact extremely difficult to hear any of the upper harmonics individually; they all seem to combine to produce a general sense of the timbre of the tone you are hearing. And varying the filter F_c seems only to change this timbre from bright to dull or somewhere in between.

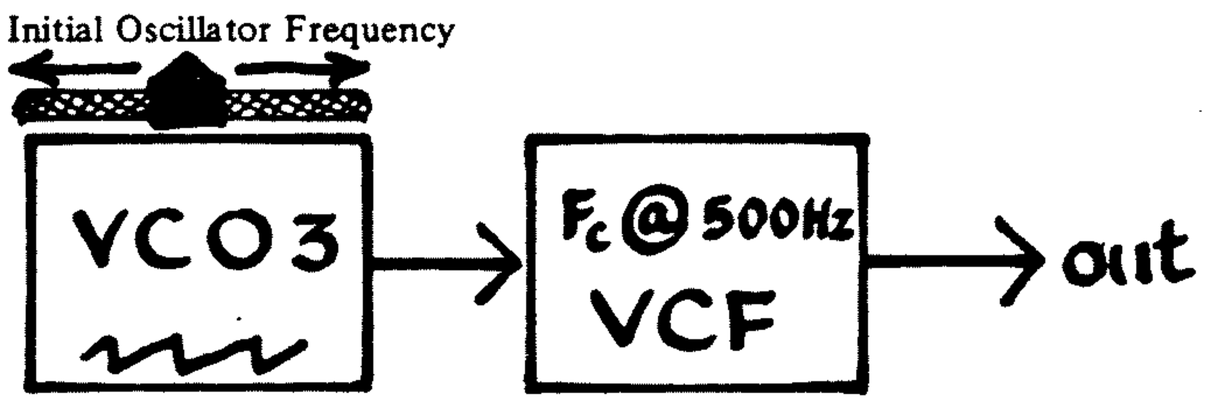
Since the resonance control is continuously variable, it can give any desired degree of peaking short of the extremely sharp peak you have just used. For musical purposes these intermediate positions are most useful; in conjunction with the variable F_c of the filter, they give you an enormous amount of control over the timbre of the musical tones you wish to produce.

For of course, as you discovered earlier, each of the five waveforms available from the VCO's has its own timbre, even without filtering. And this is the place to experiment with each one of them, following more or less the same procedure as we outlined for the sawtooth above. You will discover among other things that

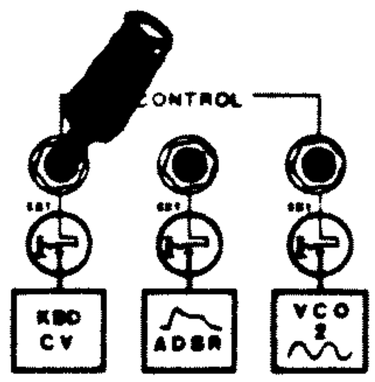
1) the square wave has only odd harmonics: fundamental, twelfth, third above the second octave, flat seventh above the second octave, and so on, representing the first, third, fifth, seventh harmonics and the others in the same series.
2) the sine wave has no harmonics: the filter output will peak once, at the frequency of the sine wave, and at no other frequencies.
3) The harmonic structure of the pulse wave depends on the width of the pulse.



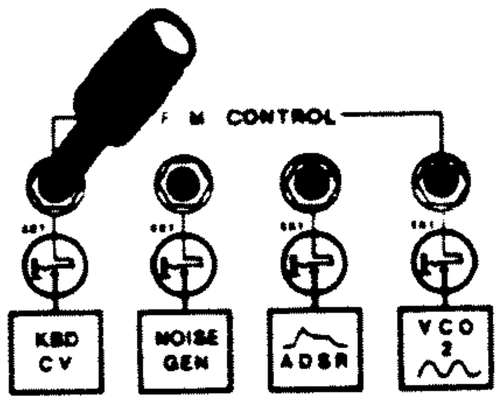
5.1072 It would be a good idea too to perform a few other experiments to familiarize yourself thoroughly with the action of the VCF and with its timbre-control possibilities. We outline them briefly here.



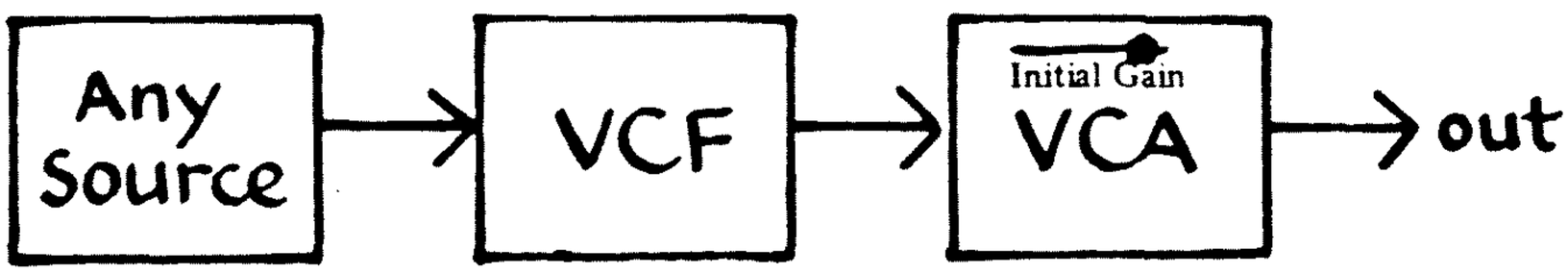
1) Instead of leaving the input frequency constant and sweeping the filter frequency, leave the filter frequency constant and sweep the input frequency.
2) Depressing the lowest key on the keyboard, tune the filter at fairly high resonance to some upper harmonic of an input waveform. Play a tune on the keyboard and note that, as long as the filter is fed with a keyboard control voltage, it will follow the harmonic you've tuned it to. Now disconnect it from the keyboard voltage by inserting a dummy plug in the KBDCV-Filter control input. Play the same tune on the keyboard again and note the difference; this time the filter "picks out" whichever harmonic component of the input signal lies closest in frequency to the peaking frequency of the filter. If any note in your tune happens to have no harmonic components near the filter F_c the filter output will be greatly minimized.



3) Disconnect VCO3 from the keyboard control voltage with a dummy plug inserted in the KBDCV-VCO3 input. Tune it to about 200Hz and then tune the VCF to an octave above it. Remember to first remove the dummy plug from the KBDCV-Filter input and to touch the lowest note on the keyboard before tuning the filter. (Note: to tune the filter to any particular pitch you can either tune it in its oscillating state and then lower the resonance, or simply tune it to some harmonic of an input signal. See section 5.1071, paragraph 2.) Now you can "play" on the harmonics of the VCO3 sawtooth by playing up and down the keyboard. In the lowest keyboard octave only the C, E, G, and B-flat will be active, corresponding to the 4th, 5th, 6th, and 7th harmonics of the sawtooth. But in the next octave all the notes will be reasonably active and reasonably close to tempered pitch except the F and A; and in the next octave still you can get all 12 notes of a chromatic scale. (There are in fact 16 harmonics in this octave.)

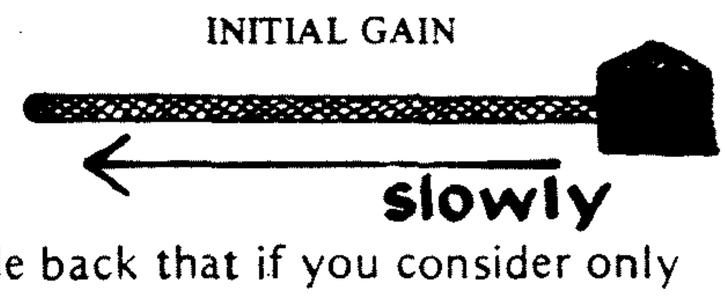


This concludes your preliminary acquaintance with the VCF.



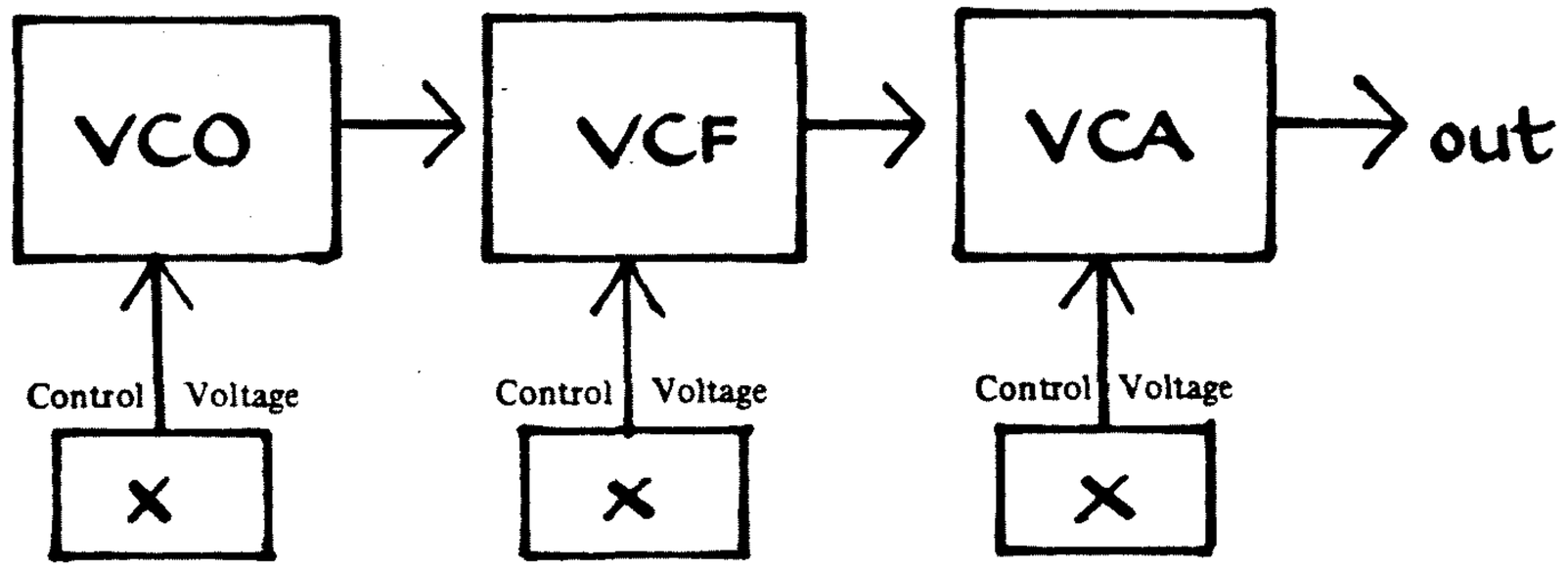
5.108 Now close the VCF-Mixer input and open the VCA-Mixer input. Open the VCF-VCA input and raise the VCA initial gain to maximum. Feed any audio signal you please into the filter; note that with the new signal path you have set up, the signal you are listening has passed from the filter *through the amplifier* before going into the mixer.

Close the VCA-Mixer input and open the VCF-Mixer input again. The two signals should be identical: the maximum output level from the VCA is no higher than the level of its input signal. Close the VCF-Filter input and open the VCA-Filter input once more, and now slowly move the VCA initial gain slider from extreme right to extreme left. The signal is slowly attenuated until nothing remains of it.



We mentioned briefly a while back that if you consider only the manual control of signal amplitude that the VCA gives you, the VCA could perfectly well be replaced by a single attenuator. You have just seen how this is true; the kind of control you have over a signal by means of the VCA initial gain slider is not different in any way from the kind of control you have by means of say the mixer input attenuators, or the VCA's two signal input attenuators themselves. The only thing that any of them can do is lower the level of a signal passing through. From beginning to end, the patch you are listening to just now has in its signal path a total of three attenuators (four if you are listening to the noise generator) not counting the speaker volume controls.

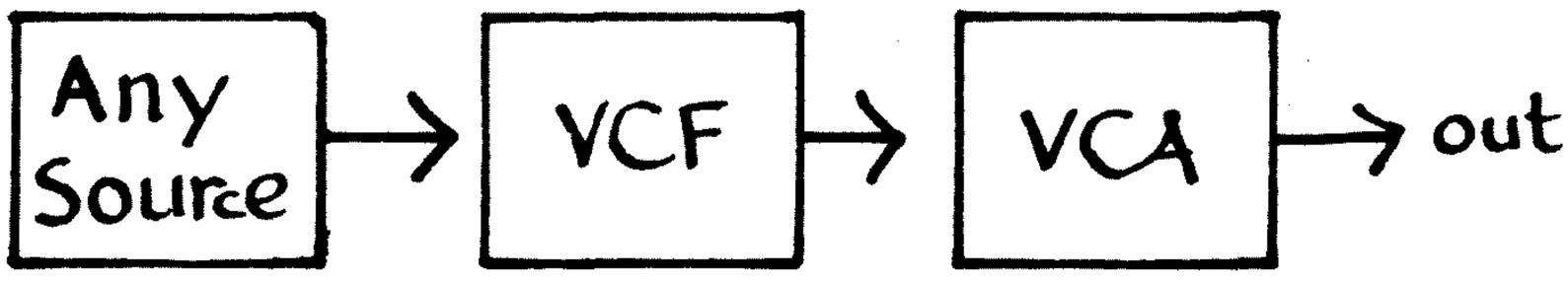
But of course the point of the VCA's presence in the 2600 is not just to give you one more manual level control. It is to give you a Voltage-Controlled amplitude-governing device. So let's control it with a voltage.



5.11 ADDING VOLTAGE CONTROL TO THE BASIC PATCH.

We have already slipped in, more or less surreptitiously, examples of how the keyboard control voltage can "play" a voltage controlled oscillator or the VCF. In this section, beginning with the patch you were using at the end of the last section, we will try to give you a little better idea of what you can do with simple forms of voltage control.

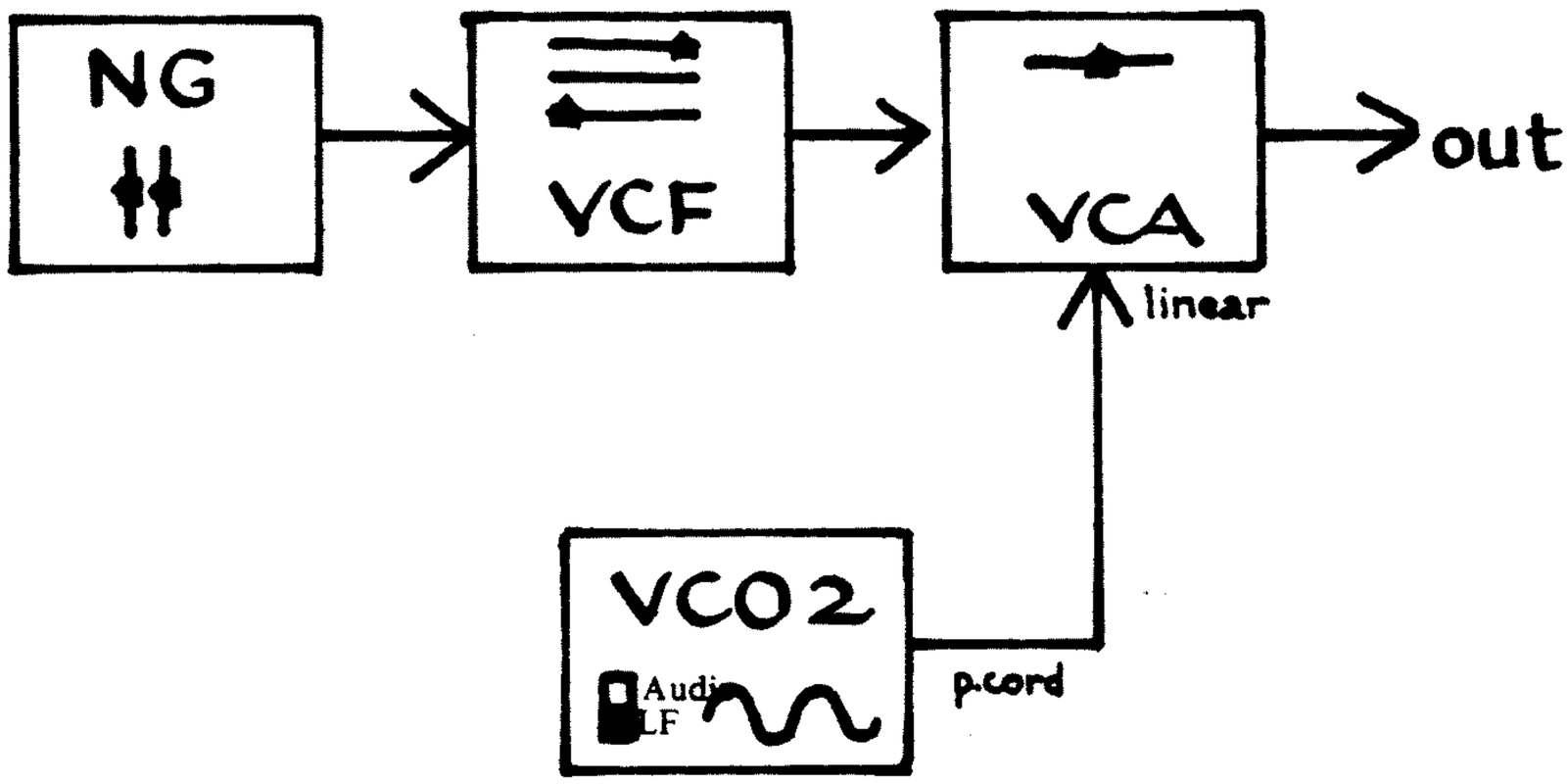
5.111 VOLTAGE CONTROL AND THE VCA. The patch you were last using was this one.



Now add one connection: take a patch cord and connect the VCO2 sine output to the linear control input of the VCA. Set VCO2 to its low-frequency range and the initial frequency slider to anywhere near center. Set the VCA initial gain about one-third up.

As your signal source use the noise generator; open the NG-Filter input and close all the other filter inputs. Set filter resonance at minimum and initial frequency at maximum.

Now open the VCA linear control input attenuator about half-way. This should introduce a slow undulation into the volume of the noise you are listening to: the slowly undulating output voltage of the VCO2 sine wave is increasing and decreasing the gain of the VCA.



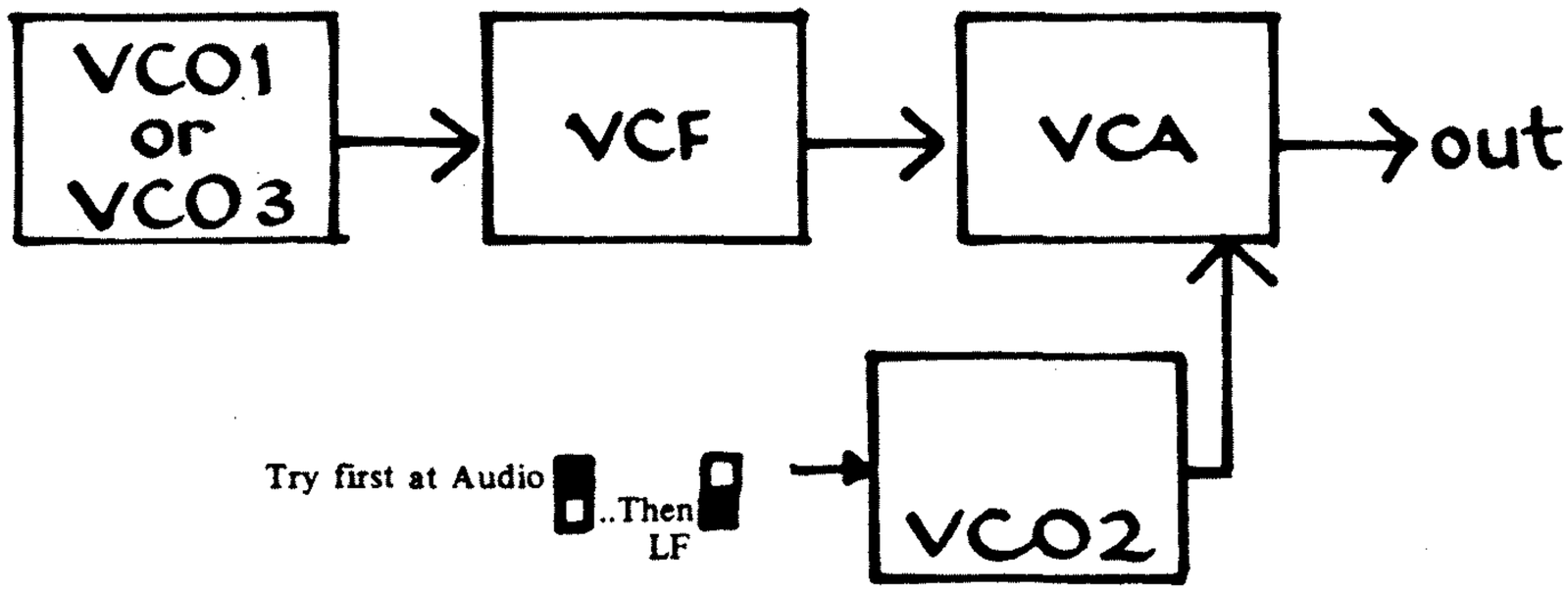
The control input attenuator, by governing the strength of the control signal allowed into the amplifier, governs the amount of control obtained. In this case, opening the attenuator all the way allows the full 10V p-p voltage of the sine wave in. This is enough to drive the VCA all the way from maximum to minimum gain. Overloading the control input, which may occur with sòme input signals, can be recognized by the discontinuities it causes in the VCA output; it does no harm and can be used deliberately if you wish. (In general this is true of all the 2600 functions.)

Since the voltage signal you are using in this particular patch originates in VCO2, it is reasonable to expect that changing the behavior of that oscillator will cause a corresponding change in the behavior of the VCA. A little fiddling with the frequency slider of VCO2 will demonstrate that this is indeed the case. Call the undulating gain of the VCA a tremolo and say to yourself that the frequency of the tremolo is directly governed by the frequency of VCO2. And of course there isn't any reason in the world why a tremolo shouldn't be able to happen not only at relatively slow rates, like those produced by VCO2 in its low range, but also at higher rates or even at audio grequencies.

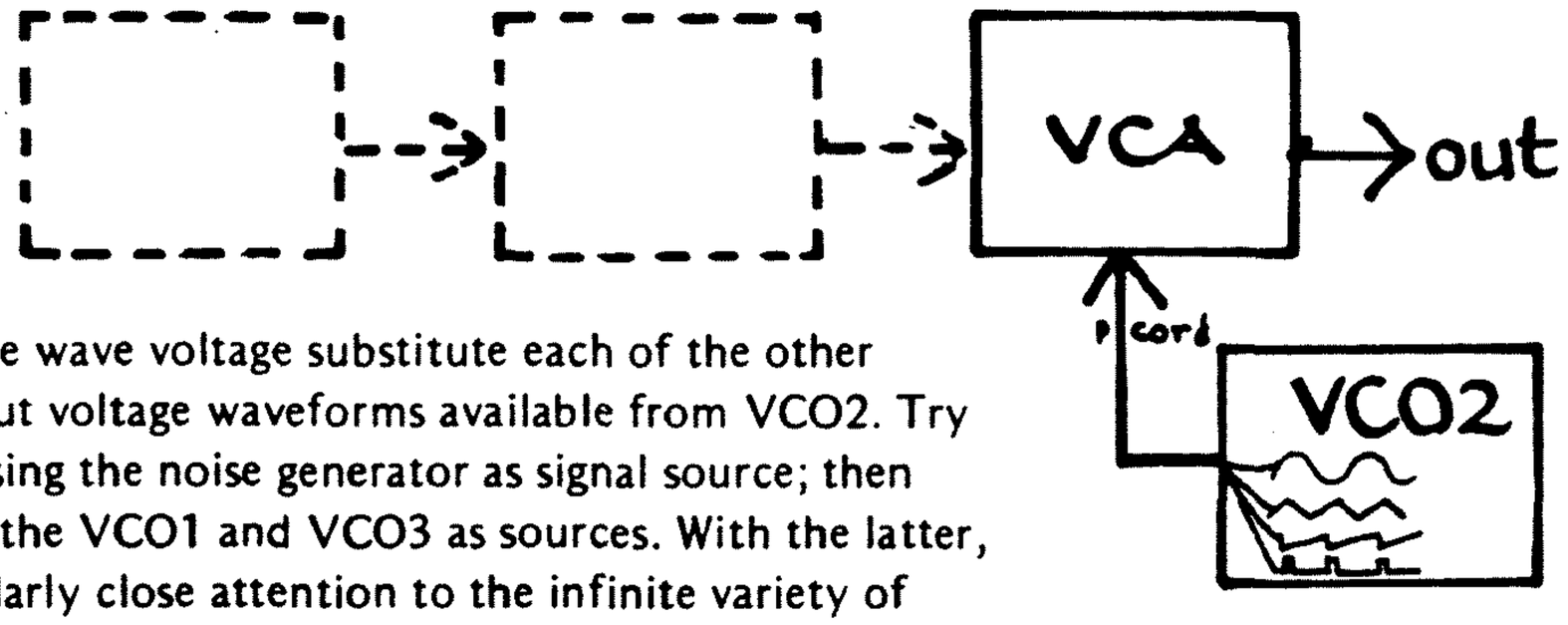
With this in mind, you might try switching VCO2 to its audio range and begin sweeping it upward from 10Hz. The resulting output from the VCA has of course no audible similarity to anything you would ordinarily call a "tremolo" at all; it is in fact one of a whole class of what we shall call "amplitude modulation effects" or simply "AM effects" for short.

In general, a fluctuating control voltage will have entirely different audible effects depending on whether it fluctuates predominantly at subsonic or at audio frequencies. And the effects it produces on VCO, VCF, or VCA behavior when it fluctuates primarily at audio frequencies we shall call specifically "modulation effects"—although, as we have tried to make clear, any systematic modification of a waveform may technically be considered a "modulation" of that waveform. The concept of modulation is sometimes even made so general that any change whatsoever in a waveshape is called a modulation; the context in which the word is used must generally be depended on to define the scope of its meaning.

5.1111 FURTHER RESEARCH.



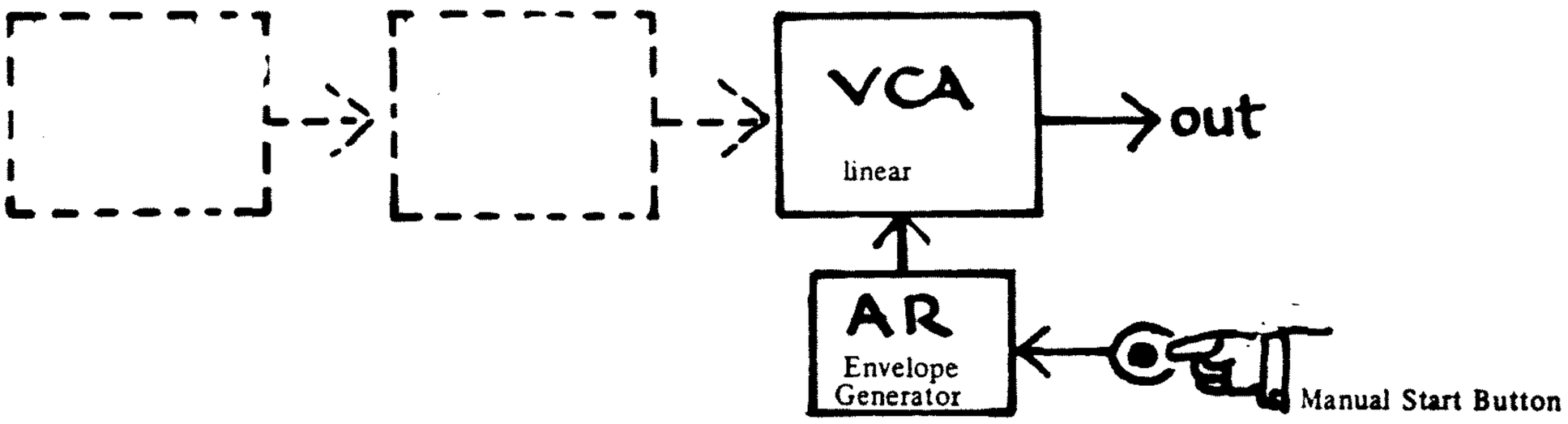
1) For the noise generator you were listening to, substitute one of the VCO's not in use—VCO1 or VCO3. Listen first with the controlling oscillator VCO2 in its low range, then in its high range. Don't forget to try different input level settings for the linear control input of the VCA.



2) For the sine wave voltage substitute each of the other three output voltage waveforms available from VCO2. Try this first using the noise generator as signal source; then try it with the VCO1 and VCO3 as sources. With the latter, pay particularly close attention to the infinite variety of AM effects produced by the interaction of the signal frequency and the control frequency. Significant differences in the VCA output will be produced by changing a) signal frequency b) control frequency c) control input level, and d) initial gain setting of the VCA.

Note that, with the VCA initial gain at minimum, a low-frequency square wave fed to the control input simply functions as a kind of gate; when the square-wave voltage goes to +10V, the amplifier is driven upto its maximum output, and when the voltage returns to zero the amplifier is effectively shut off. Attenuating this input voltage lowers the output level of the amplifier.

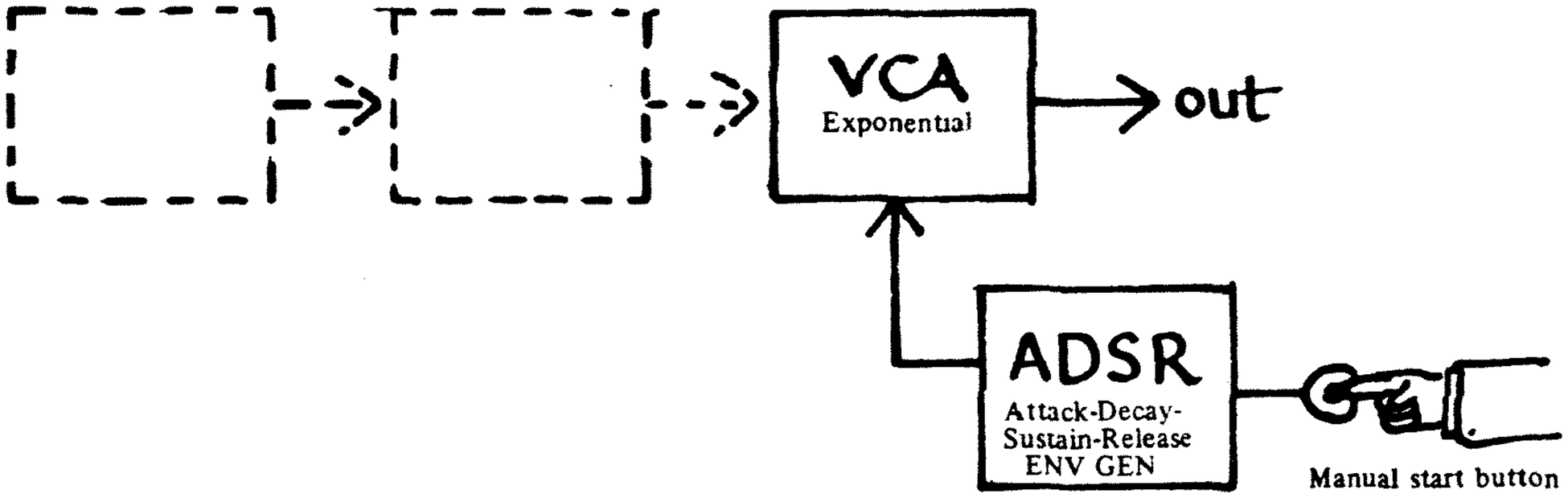
With this in mind, remove entirely the patch-cord connection to the VCA linear control input and try



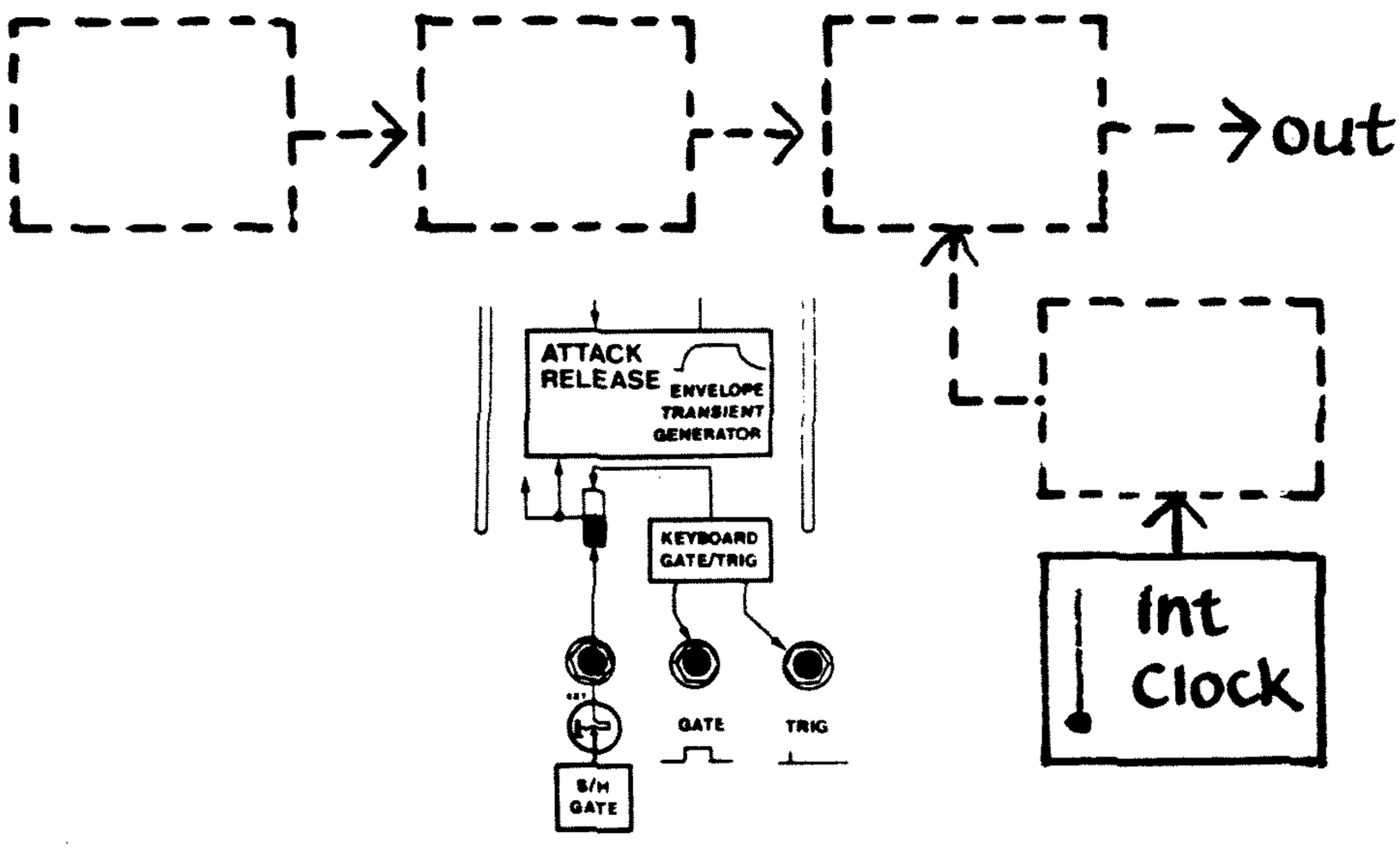
5.1112 USING THE ENVELOPE-GENERATOR CONTROL VOLTAGE TO CONTROL THE VCA. Use any signal input to the VCA. Set initial gain to minimum and open the linear control input to maximum. The prewired connection to this input is from the AR Envelope Generator. With the Attack and Release control sliders on this generator both at minimum, press the "manual start" button. Release it again. The envelope generator output voltage rose to +10V the instant you pressed the button and fell to zero again the instant you released the button. You should have heard the VCA output rise to maximum and fall to minimum. If you didn't, check your patch and control settings.

Raise the Release control slider to maximum and press the button again. This time the output voltage still rises instantly, and so, consequently, does the VCA gain. But when you release the button the fall in output is very slow. Raising the Attack control slider to maximum will have the effect, as you might have guessed, of introducing a rather long rise time into the generator output voltage and so into the VCA signal output. Experiment with intermediate settings of these two controls and with longer and shorter hold-down times for the manual start button. You may find a minimum attack time and maximum release time

particularly interesting with very short button-taps. The effect is of course to create a percussive kind of envelope resembling various familiar percussive sounds depending on what signal you are feeding into the VCA.

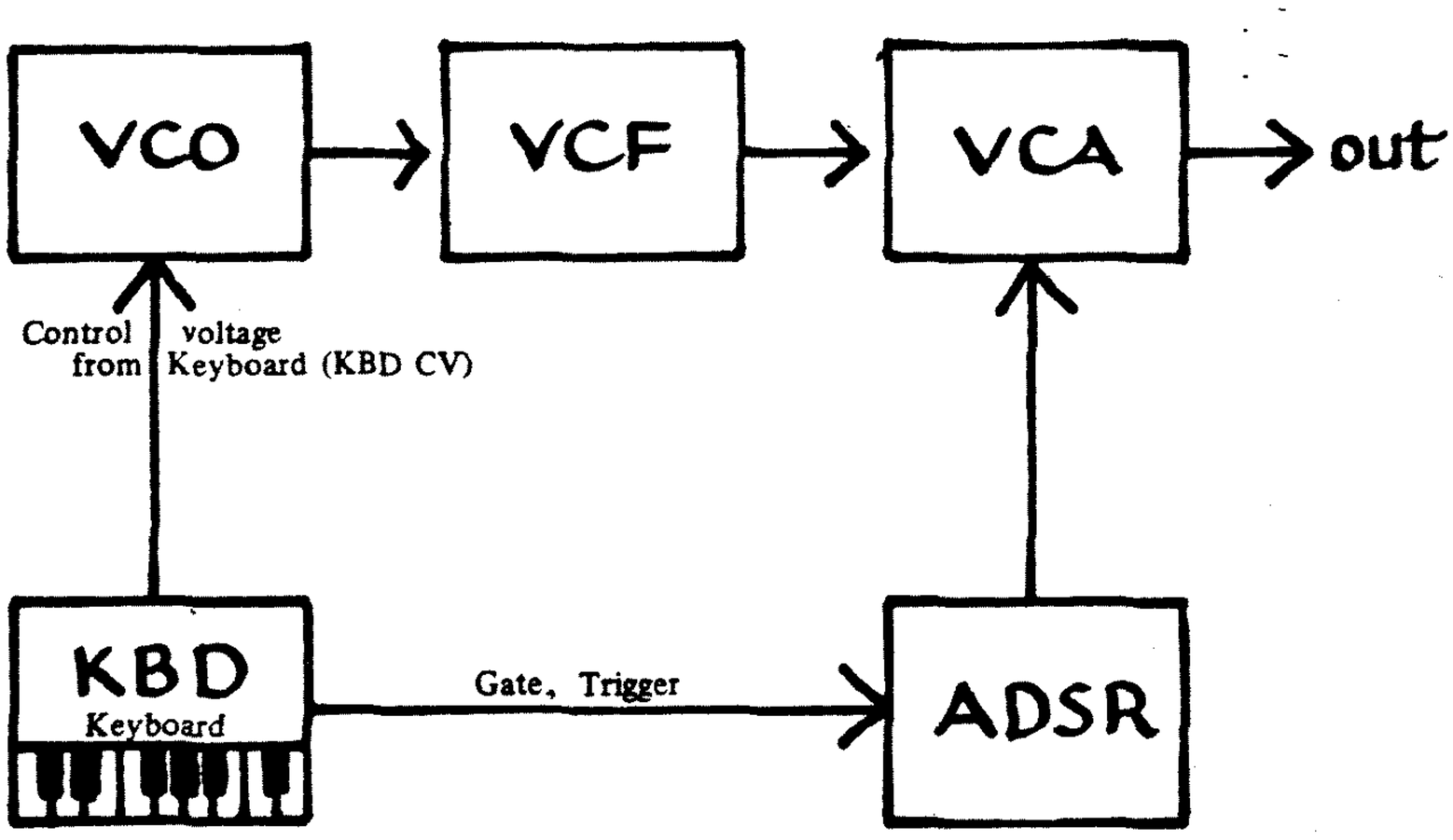


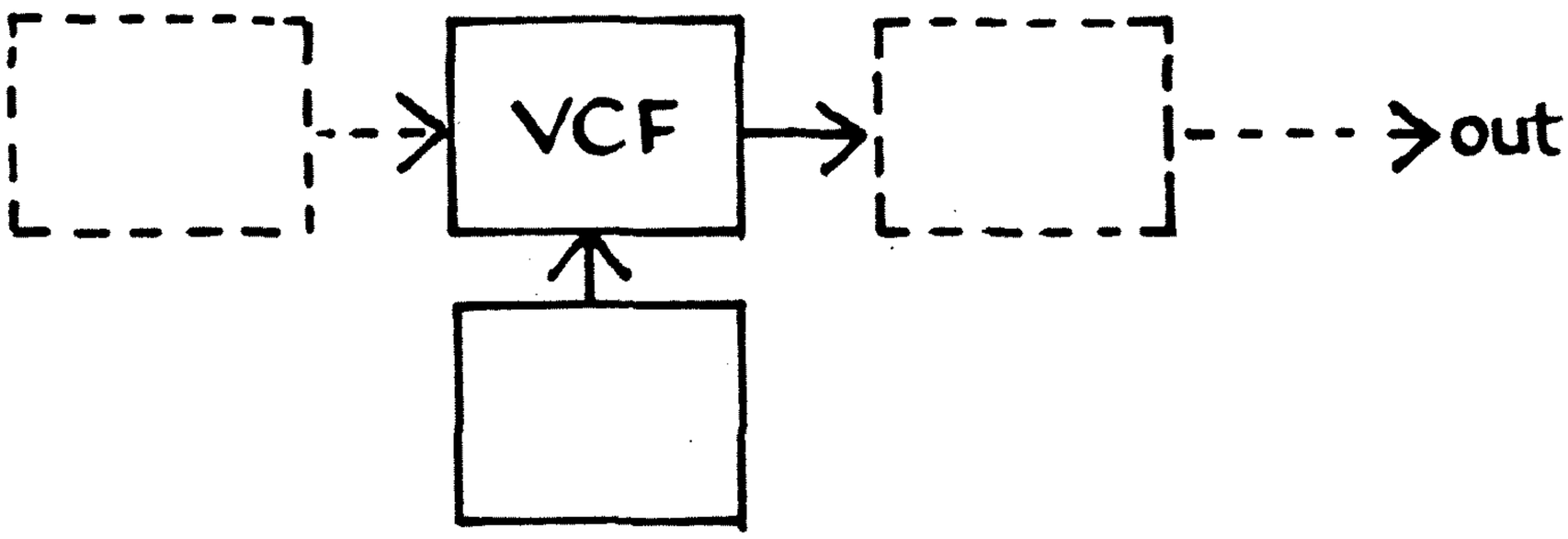
When you have exhausted the immediate possibilities of the AR envelope generator control, close the linear control input to the VCA and open the exponential input to maximum; the prewired connection to this input is from the ADSR Envelope Generator. You can start all over again. If you get tired of jabbing the manual start button you may flick the gate selector switch to its lower position, in which the envelope generators are gated by an external input which is prewired from the Sample & Hold. This way the envelope generators will produce an envelope automatically every five seconds or so at the lowest setting of the Internal Clock "rate" control.



The ADSR generator can produce somewhat more complex envelopes than the AR generator, and it will be well to familiarize yourself thoroughly with them in as much detail and variety as possible.

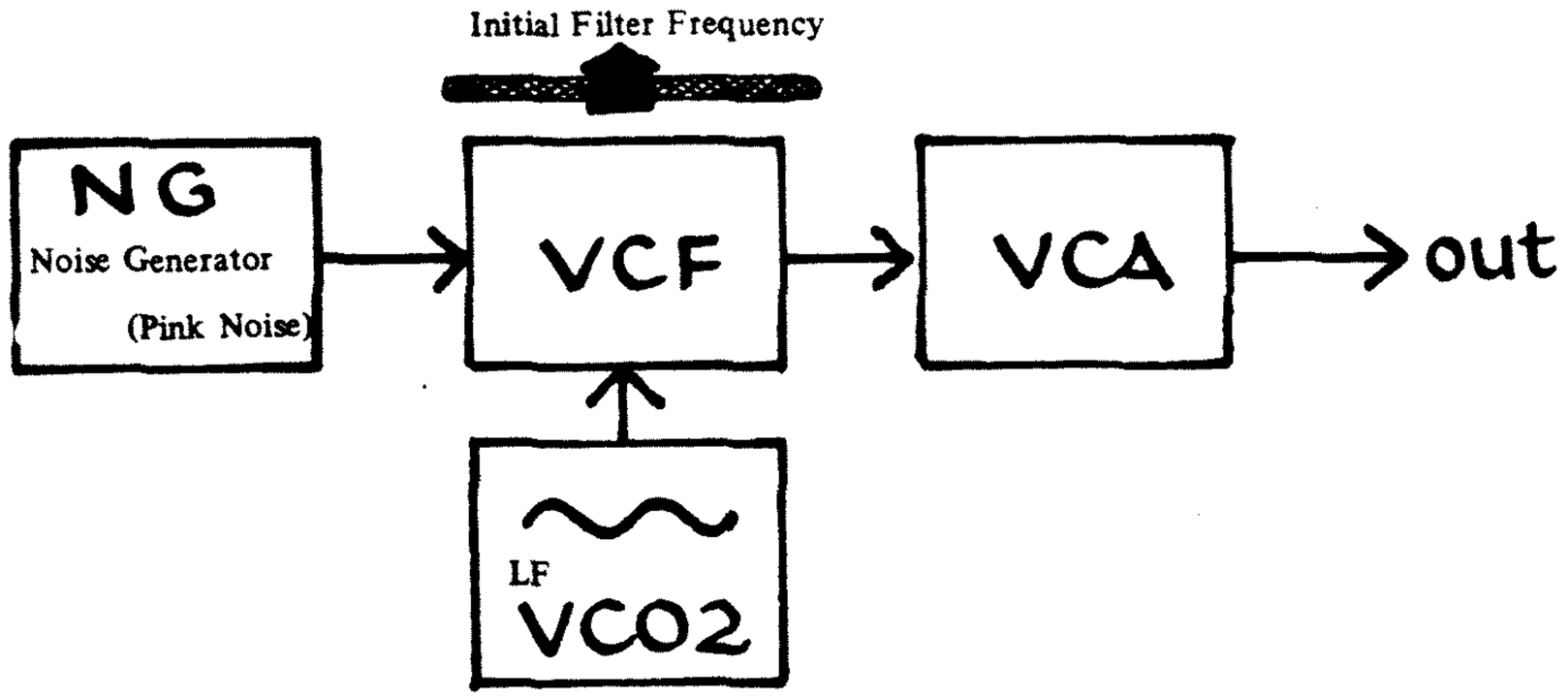
At some point in this experimentation, switch the generators from the internal clock gate back to the keyboard gate and trigger. Now feeding any oscillator output, at some audio frequency, into the filter, and from the filter into the VCA, will enable you to produce sequences of melodic events by playing the keyboard. The keyboard control voltage governs the pitch of the VCO and the keyboard gate and trigger signals operate the envelope generator.





5.112 VOLTAGE CONTROLLING THE VOLTAGE CONTROLLED FILTER.

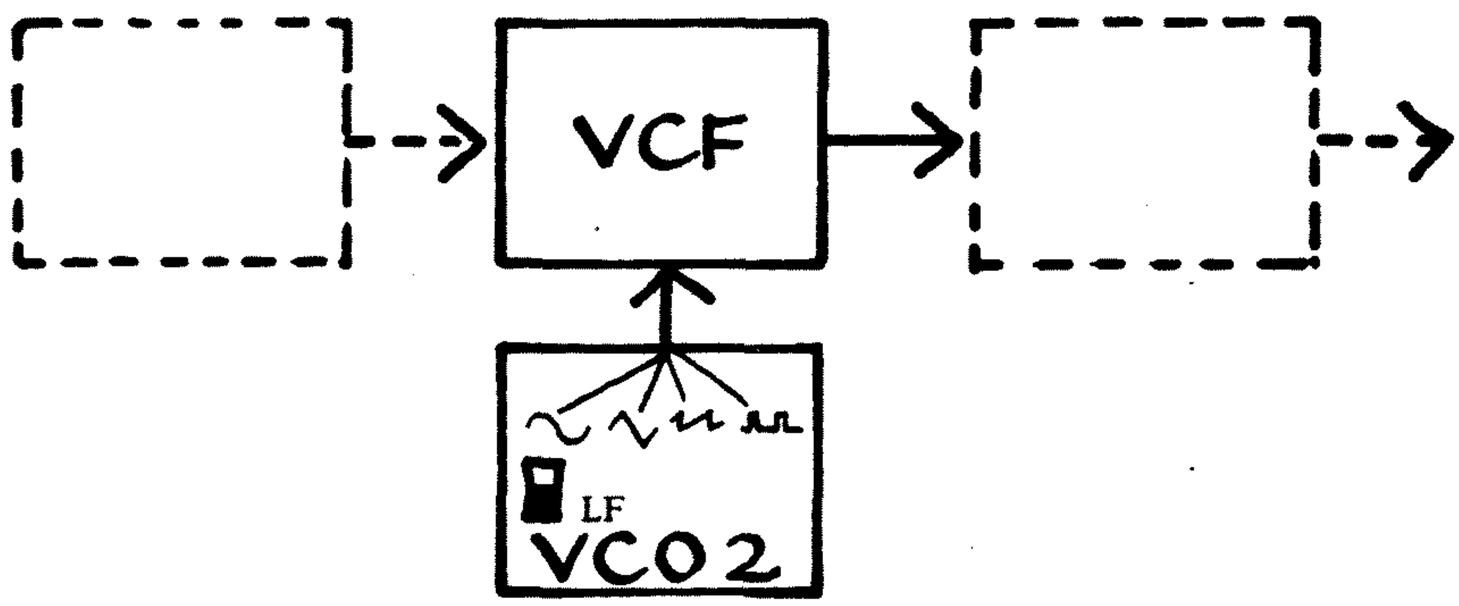
Follow pretty much the same sequence as you did with the VCA. If you begin with the VCO2 sine wave you won't need to use a patch cord; it's prewired to one of the VCF control inputs. Your best bet for an input signal is to use the noise generator. A complete statement of the patch might go like this: NG-Filter input open, VCF-Mixer input open, all other input attenuators closed. This is the signal path. The control path is provided simply by opening partially the VCO2-Filter control input (not the VCO2-Filter signal input). And the control settings are: NG color control centered, output level ¾ open; VCF initial frequency 500Hz, resonance minimum; VCO2, your control-voltage source, in its low-frequency range with frequency sliders centered.

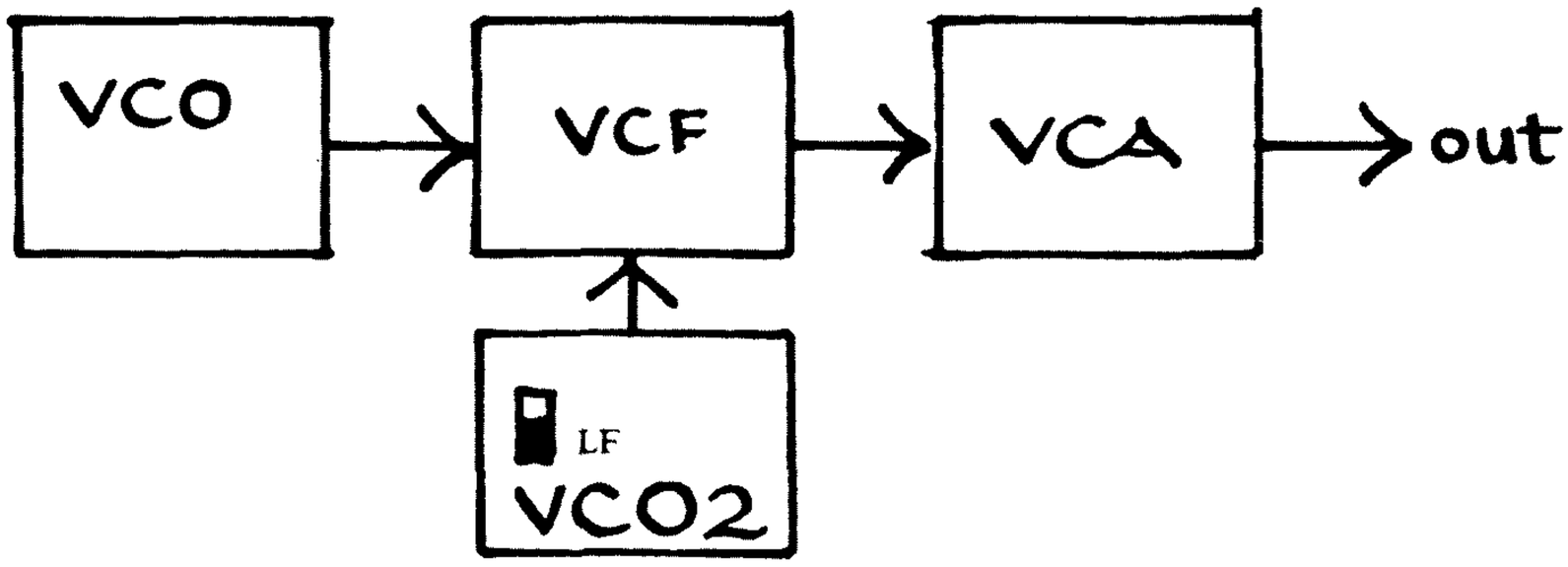


(That's a complete description of the patch. There are briefer ways of describing something like this, which will become more and more useful to you as you accumulate more and more mental baggage in your work with the 2600. Here's one way. "Pink noise is passed through a low-pass filter of F_c 440 and Q 4. F_c is modulated by a sine wave at 6 Hz to a depth of $\pm$ 1/12 octave(s), or a factor of (x times F_c)." It's all there, as you'll find by reading it carefully. Only a few things need to be commented on. 1) "Q" is an abbreviation for resonance. It can be given a numerical value but you needn't do that; just "low-Q", "medium-Q", and "hi-Q" will probably do for a long time. 2) "depth" refers to the maximum deviation (from an initial setting) caused by the control voltage. It's a convenient term for by-passing the need for talking about the strength of the control voltage itself, or even referring to it at all. In referring to the "depth" of modulation, you are talking only of the effects of a control voltage.)

Experiment, then, using just the sine wave control, with different degrees of modulation depth, different initial-frequency settings, different amounts of resonance up to and including oscillation, and different modulation frequencies from VCO2.

Then follow the same sequence with the other waveforms from VCO2, using patch cords to feed them into the VCO2-Filter control input.

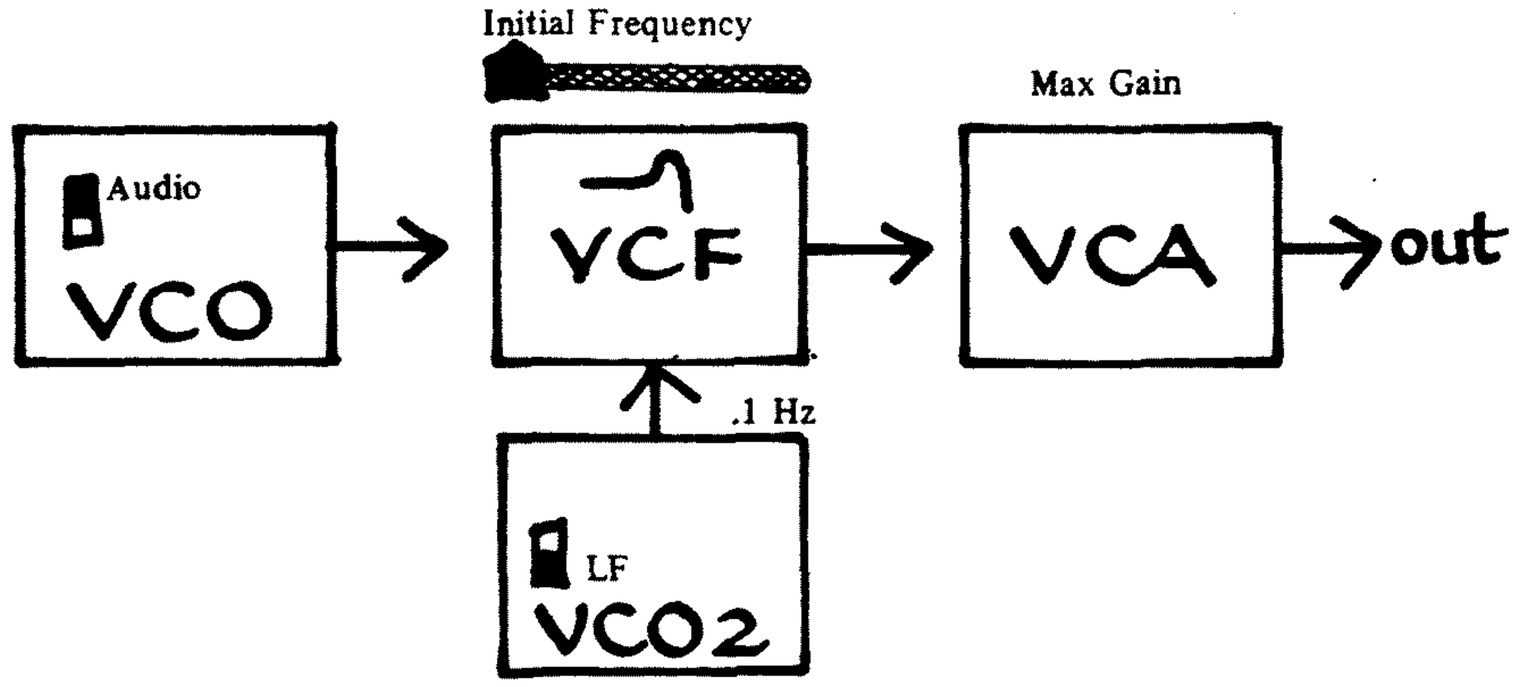




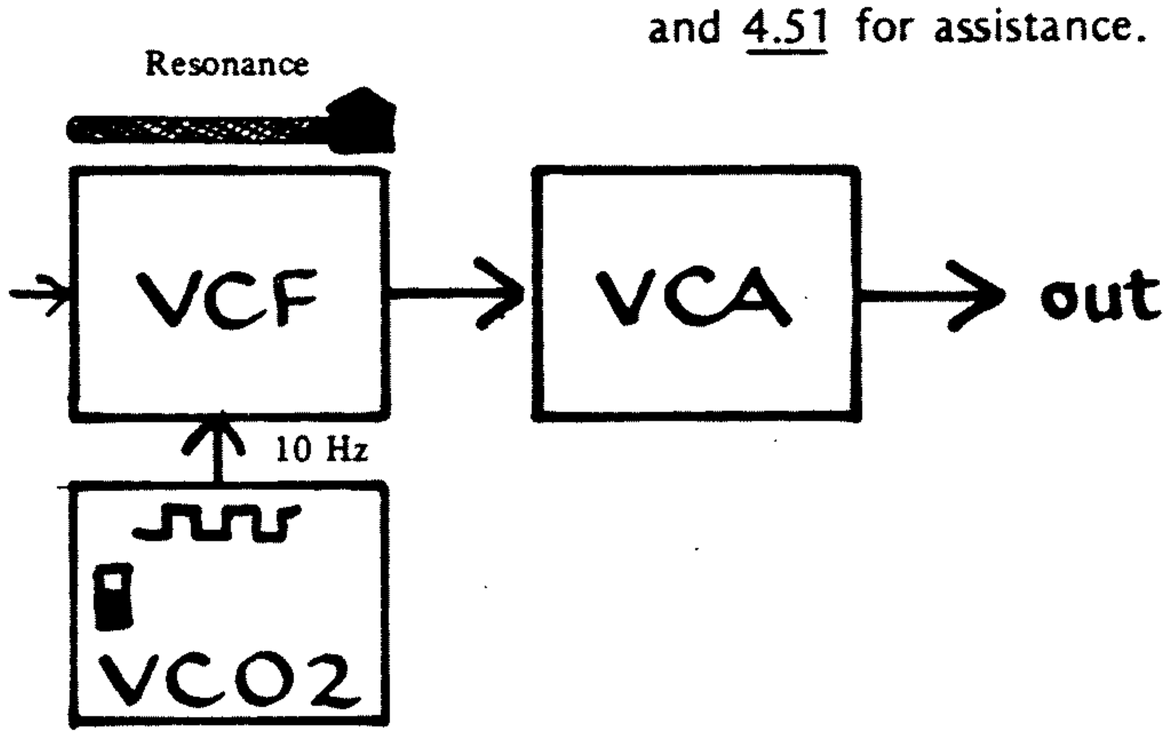
So far you've been using only pink noise as your sound source. Now close the input from the noise generator and try instead feeding an audio signal from VCO1 or VCO3 into the filter. Again take special note of the very general and basic difference between low-frequency modulation of the filter F_c and audio-frequency modulation of the filter F_c.

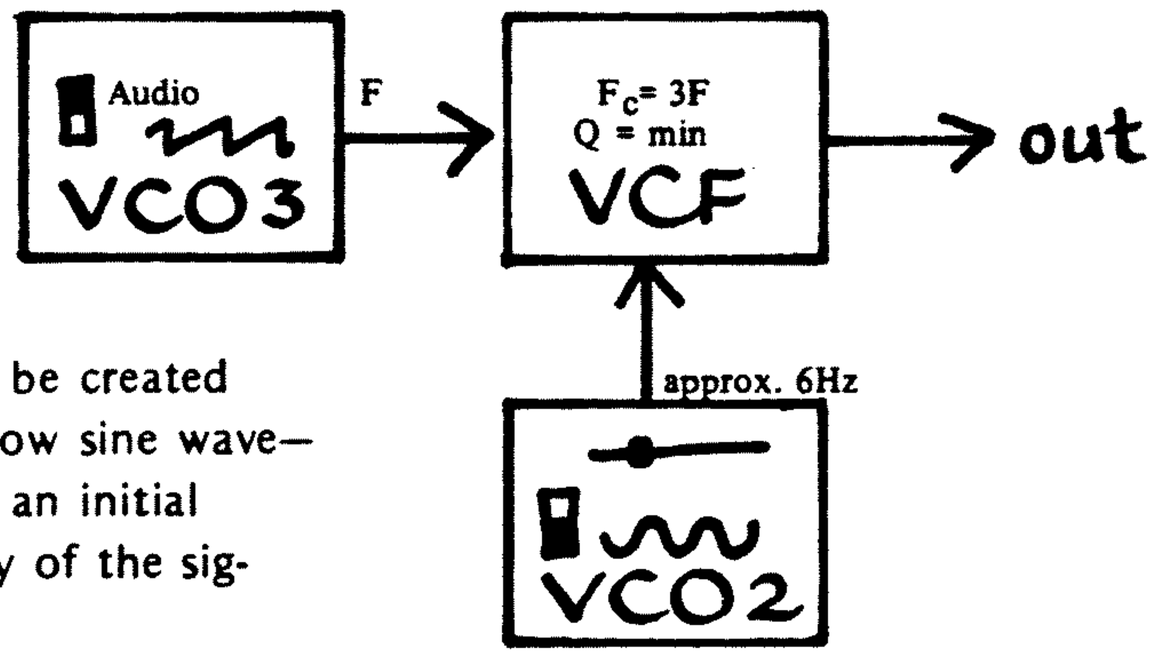
Repeat, with the signals from VCO1 and then VCO3, the same series of investigations you performed with pink noise. Here are a few more or less random observations on some of the things to take note of.

1) With the VCF just below oscillation, and any periodic waveform as a signal input, a slow sawtooth will sweep the filter automatically through the harmonics of the input waveform. (An automatic version of the experiment you performed in 5.1071.)

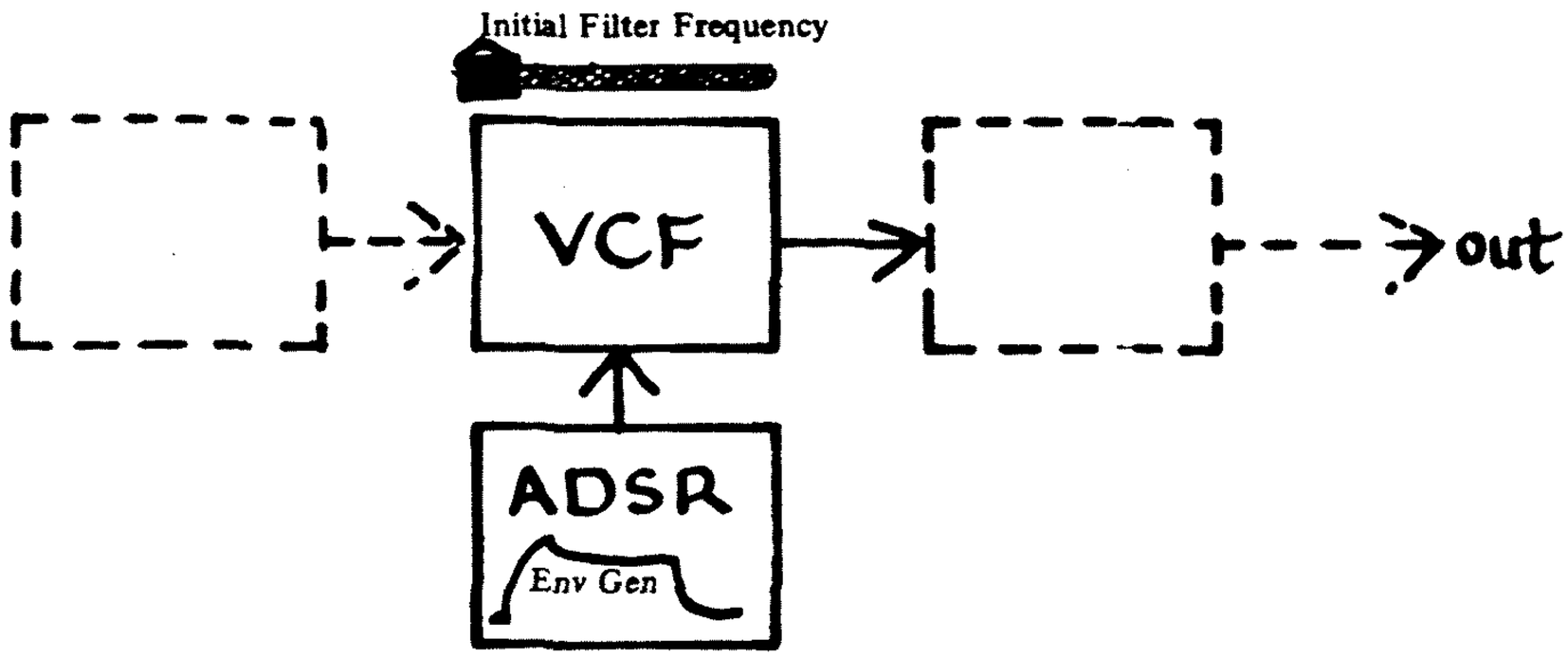


2) Square-wave modulation will cause the filter to alternate between two F_c's. Careful adjustment of the modulation depth will set the higher of these to any particular frequency you choose. The lower one will remain constant at the initial frequency setting of the VCF slider. Just for fun, patch the VCO2 square wave into the No.1 input to the voltage processor, the processor output (indicated by the panel graphics) into the control input you've been using on the filter; note that now the modulation depth affects the lower of the two frequencies while the upper one remains at the setting of the initial frequency slider. It will be worth while to spend a little time figuring out why this happens; refer to 4.112 and 4.51 for assistance.





3) Some sumptuous natural vibrato effects can be created by modulating the filter frequency with a slow sine wave—say around 6Hz—to a very slight depth. Use an initial F_c two or three octaves above the frequency of the signal being passed through the filter.



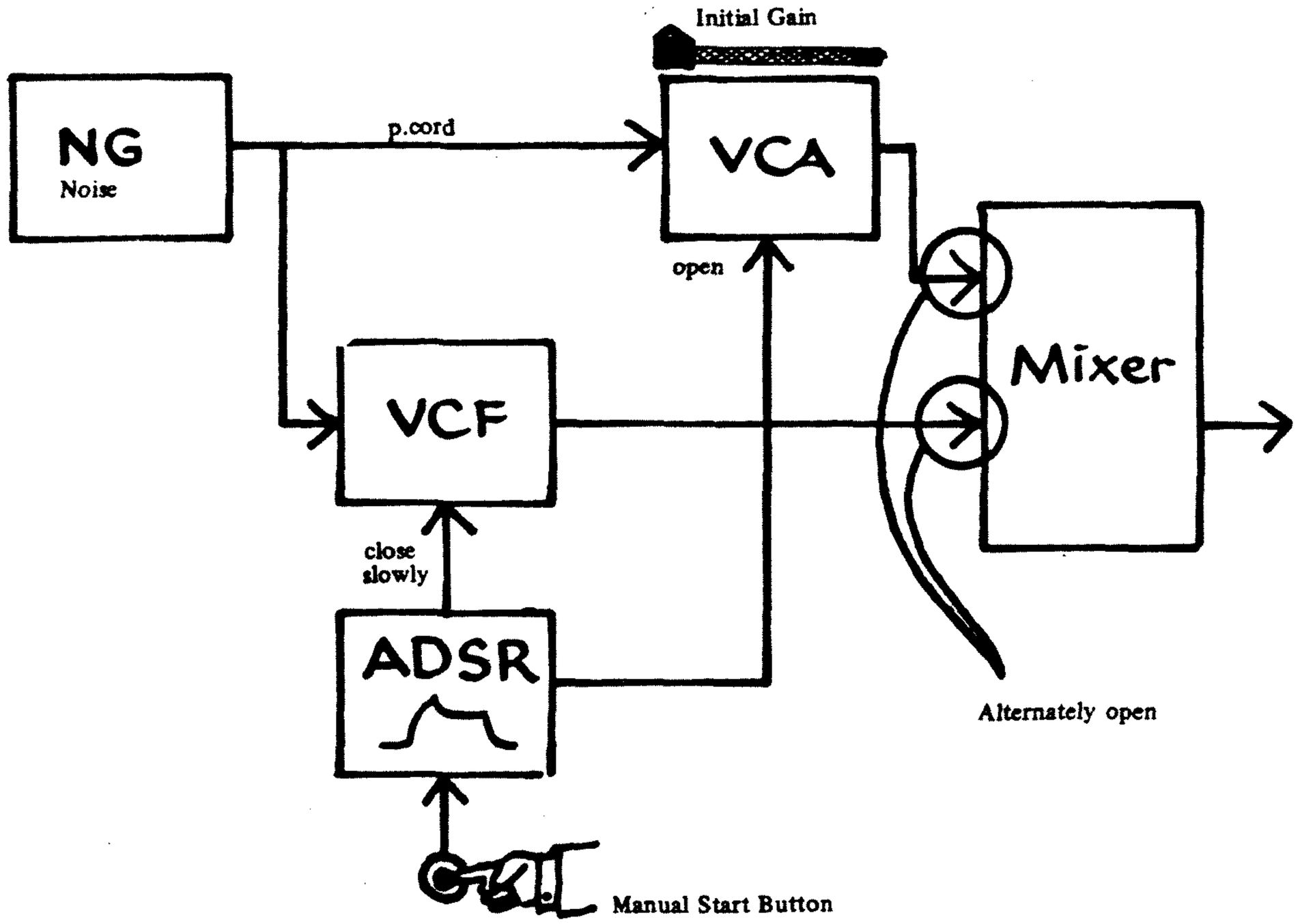
5.1121 USING THE ADSR ENVELOPE VOLTAGE TO CONTROL THE VOLTAGE CONTROLLED FILTER.

Follow the same plan of attack here that you used with the VCA, using first a pink-noise source and then one of the VCO's. Feed the VCF output directly to the Mixer; the point of our first exercise here will be to demonstrate that the VCF can create an event all by itself, without the help of the VCA.

To do this we will make use of the fact that when the filter *initial frequency control is at minimum, the filter* F_c *is so low that* the filter will not pass any audio-frequency signals. So set the initial frequency slider all the way to the left. Open the NG-Filter input all the way and open the VCF-Mixer input. Open the ADSR-Filter control input all the way. Set the ADSR envelope generator controls for a percussive short-duration envelope—say minimum attack, the initial decay about halfway up, minimum sustain level and final release.

Jab the manual start button a few times or use the keyboard gate by pressing a key. Compare this envelope with the one that results from feeding the same ADSR output and noise into the VCA.

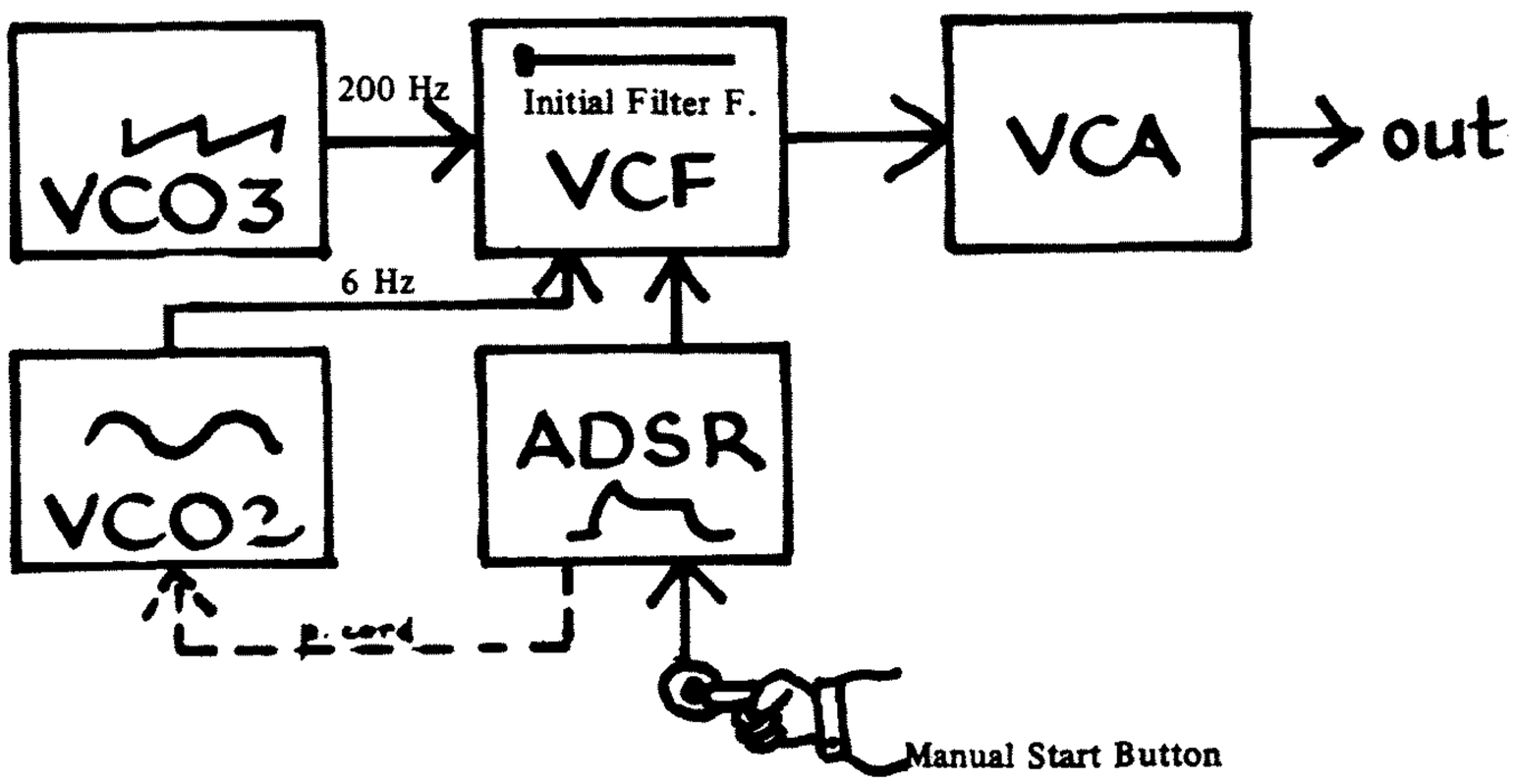
Raise the sustain level about halfway and make the comparison again. (The simplest way to make a quick patch for purposes of comparison is to take a patch cord and connect the NG output directly to one of the VCA signal inputs; open that input attenuator and the one over the ADSR Amplifier control input; now you can *simply alternate between the two mixer inputs to hear first the* VCF output and then the VCA output.)



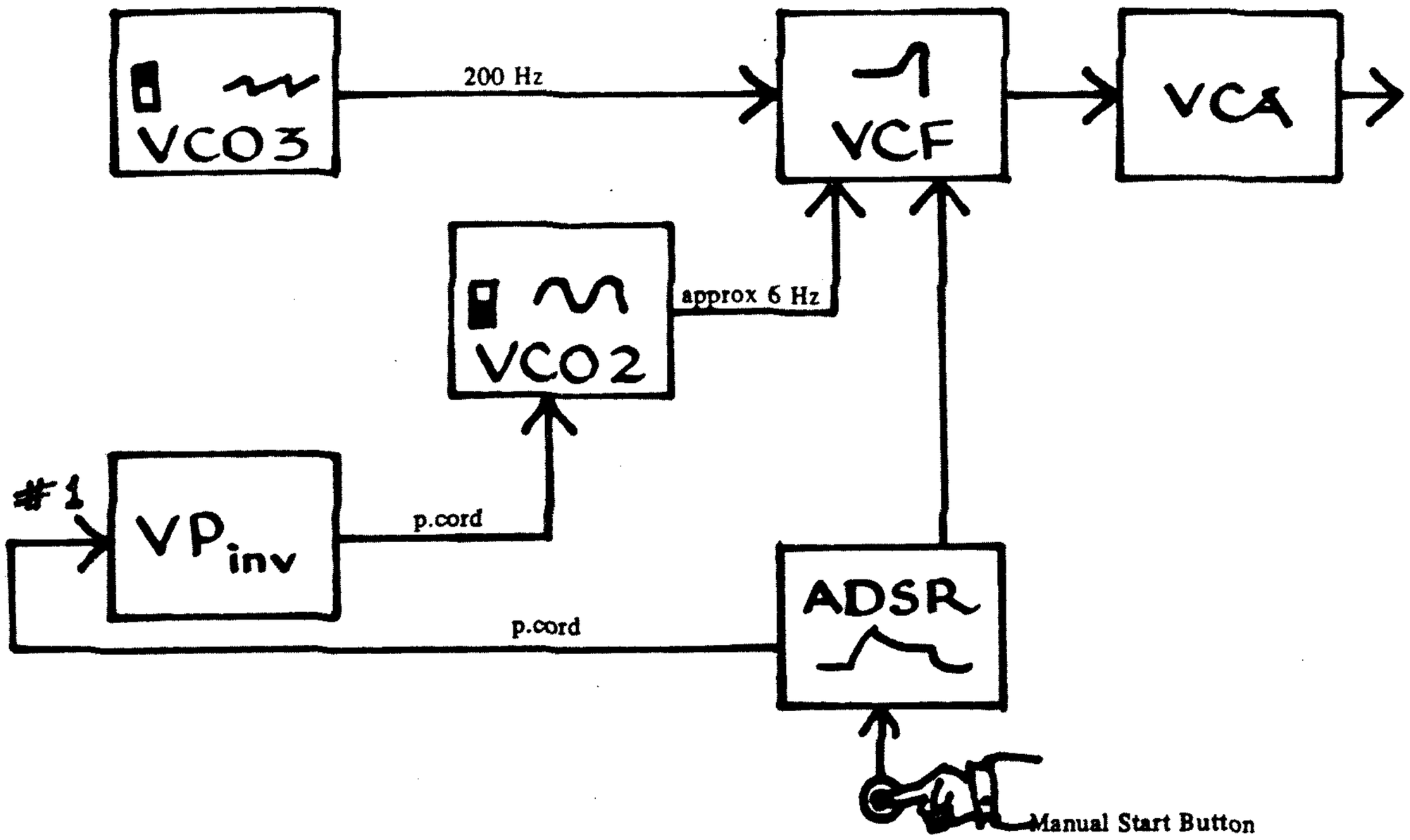
Now lower the ADSR-Filter control input attenuator to about nine-tenths of maximum and listen to the result when the ADSR generator is activated. Again at eight-tenths, and so on. As the input control voltage is attenuated, the filter is swept up to less and less of its full range and the audible output grows correspondingly duller. This is very useful technique in simulating various highly damped percussion effects such as string pizzicatos from violins, cellos, and basses.

When the resonance control is set fairly high, so that the filter peaks at F_C, you can hear vividly the changing response characteristics of the filter as the input voltage from the ADSR generator rises and falls. You have heard this sort of wah-wah before, when you experimented with other control voltages and the VCF; but they were never synchronized with an envelope. Try adding a small amount of control from the VCO2 sine at about 6-8Hz. Now you have a vibrato in addition to the envelope control. If you open very slightly the ADSR-VCO2 control slider, you will have a vibrato that changes its speed with the envelope; the ADSR voltage is not only sweeping the filter frequency, but also changing slightly the frequency of VCO2.

Under direct control from the ADSR voltage, this change in the rate of the vibrato can only be directly related to the filter F_C. In other words, the higher the instantaneous voltage from the envelope generator, the higher will be both the filter F_C and the vibrato-frequency from VCO2. But suppose you take a patch cord from the ADSR generator output and plug it into input No. 1 of the voltage inverter, and with another patch cord feed the inverter output to the ADSR-VCO2 control input. By inverting the ADSR voltage to the VCO, you will have inverted its effect, and now the vibrato rate will be slower as the filter F_C rises. In either case, the amount of change in the rate of the vibrato will be governed by the amount of ADSR output voltage that is allowed into the VCO2 input—in other words by the attenuator over that input.

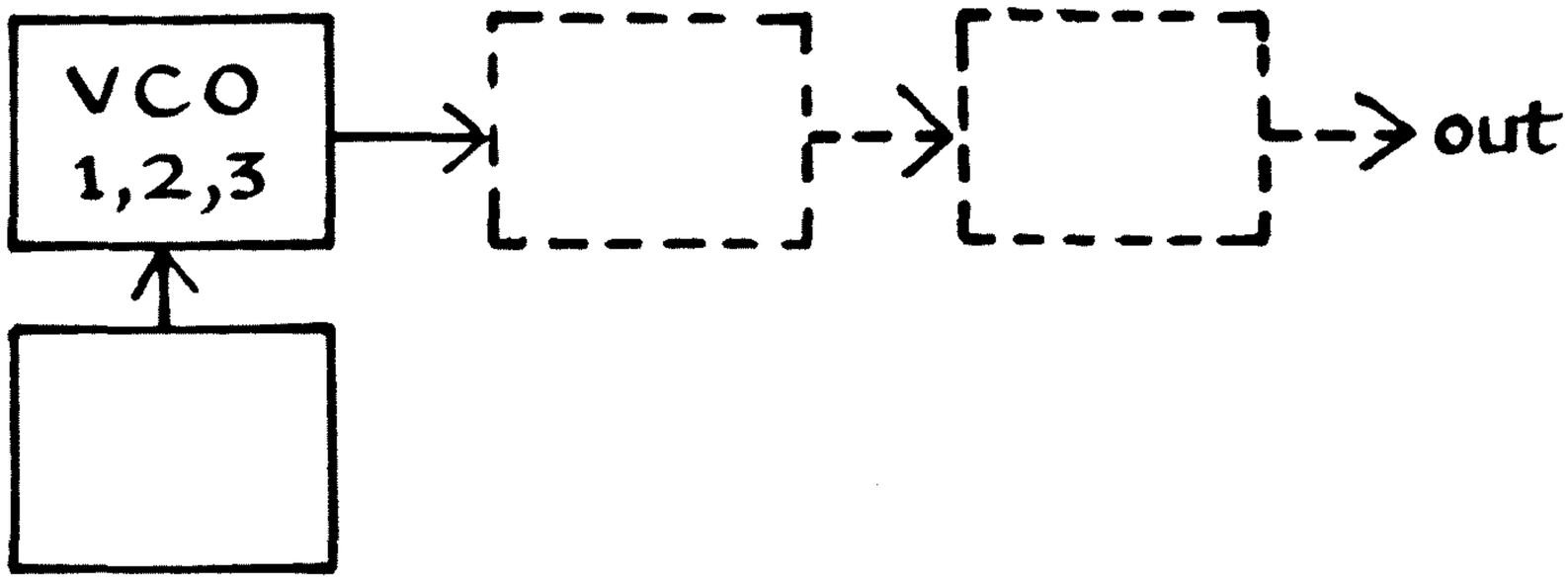


At this stage a complete block diagram of the patch would look like this.



5.113 USING CONTROL VOLTAGES TO CONTROL THE VOLTAGE CONTROLLED OSCILLATORS.

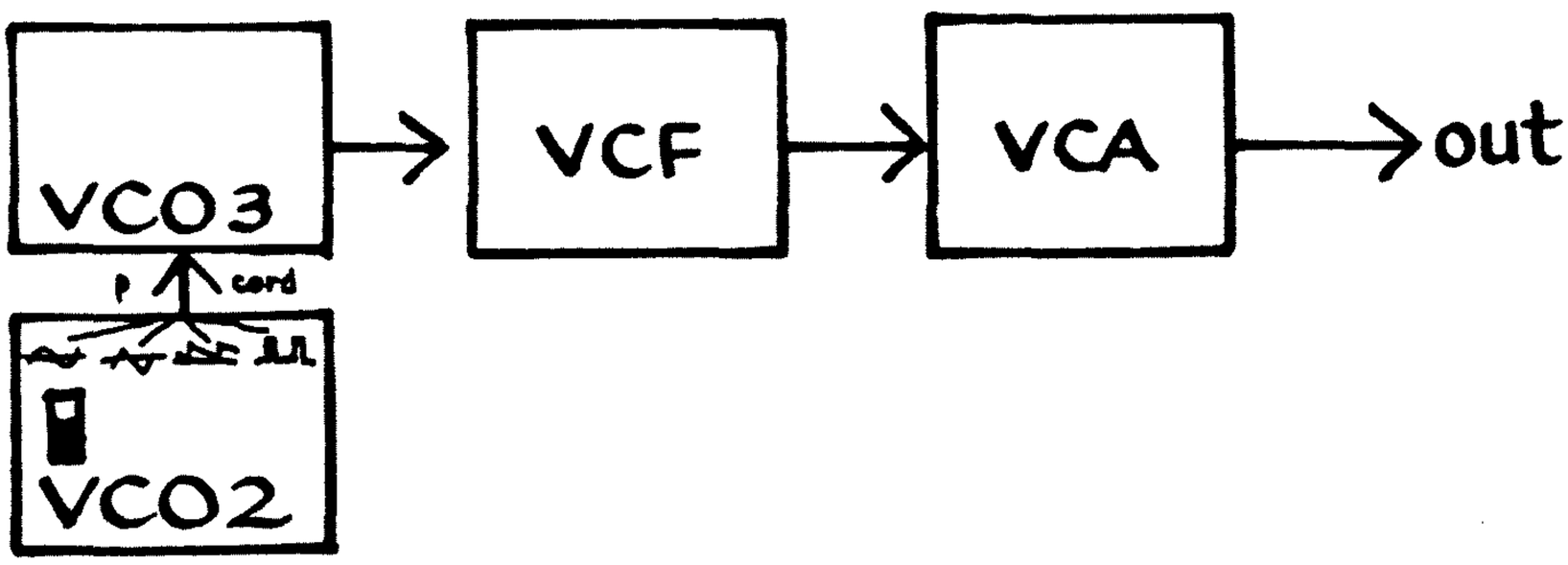
You are already familiar with the general effects possible here, from having experimented with control of the VCF in its oscillating state. Using an audio-frequency output from one oscillator to control the output of another can give rise to an enormous variety of frequency-modulation effects that are almost impossible to catalogue, and we certainly don't intend to attempt any such catalogue in these pages.



Nor is it necessary for you to spend more than a few minutes in making acquaintance with the possibilities inherent in low-frequency control. You are familiar, by this time, with patch cords, with distinguishing input jacks from output jacks, and with finding your way around the control panel. You should be able to guess pretty accurately what effects can be generated from connecting any of the functions we've looked at in detail—including the inverting voltage processor which was introduced without comment in 5.1121.

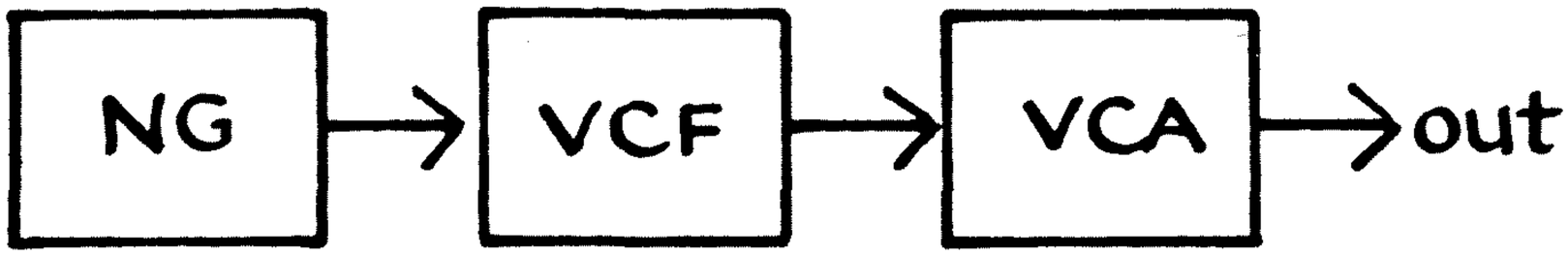
So, using VCO2 as your control voltage source (since it has all of the output waveforms at once), go through once the systematic procedure you are familiar with from the preceding sections of this chapter, beginning with low-frequency control by each waveform and then covering audio-frequency control.

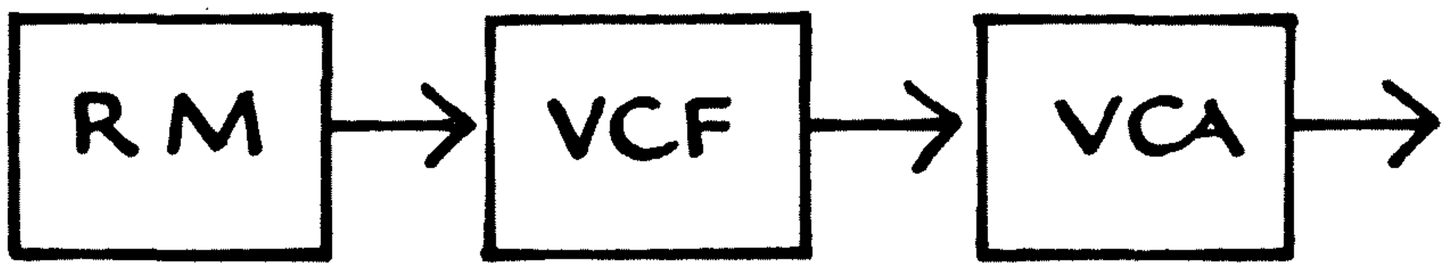
Use VCO3 as the subject of the experiments, since the pulse wave at 50% duty cycle is identical to the square wave on VCO1.



5.12 NG-VCF-VCA

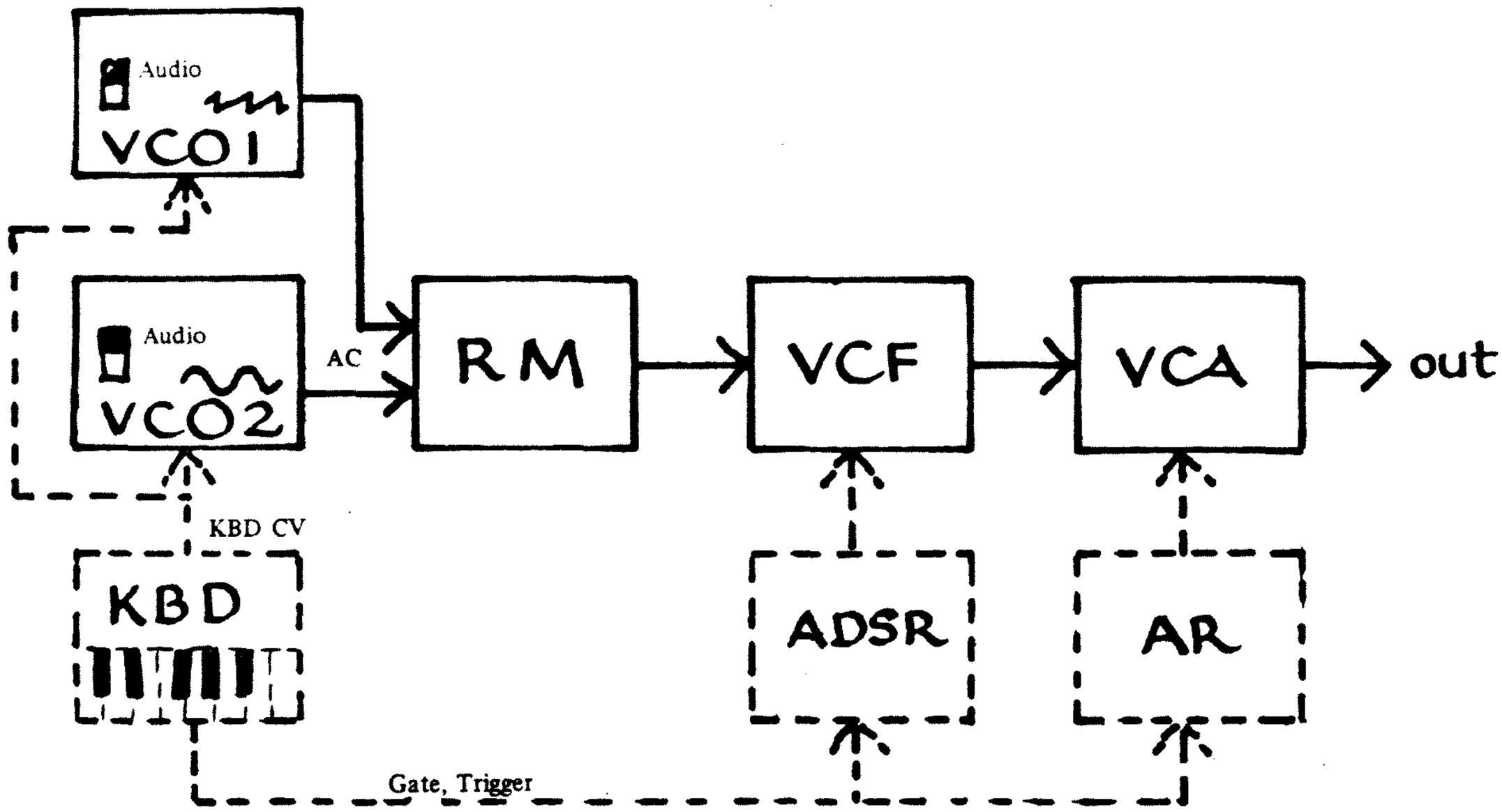
You're already familiar with this patch. We give it a separate numbering here so that you have a place to file notes on particular control settings and control voltages you find useful.





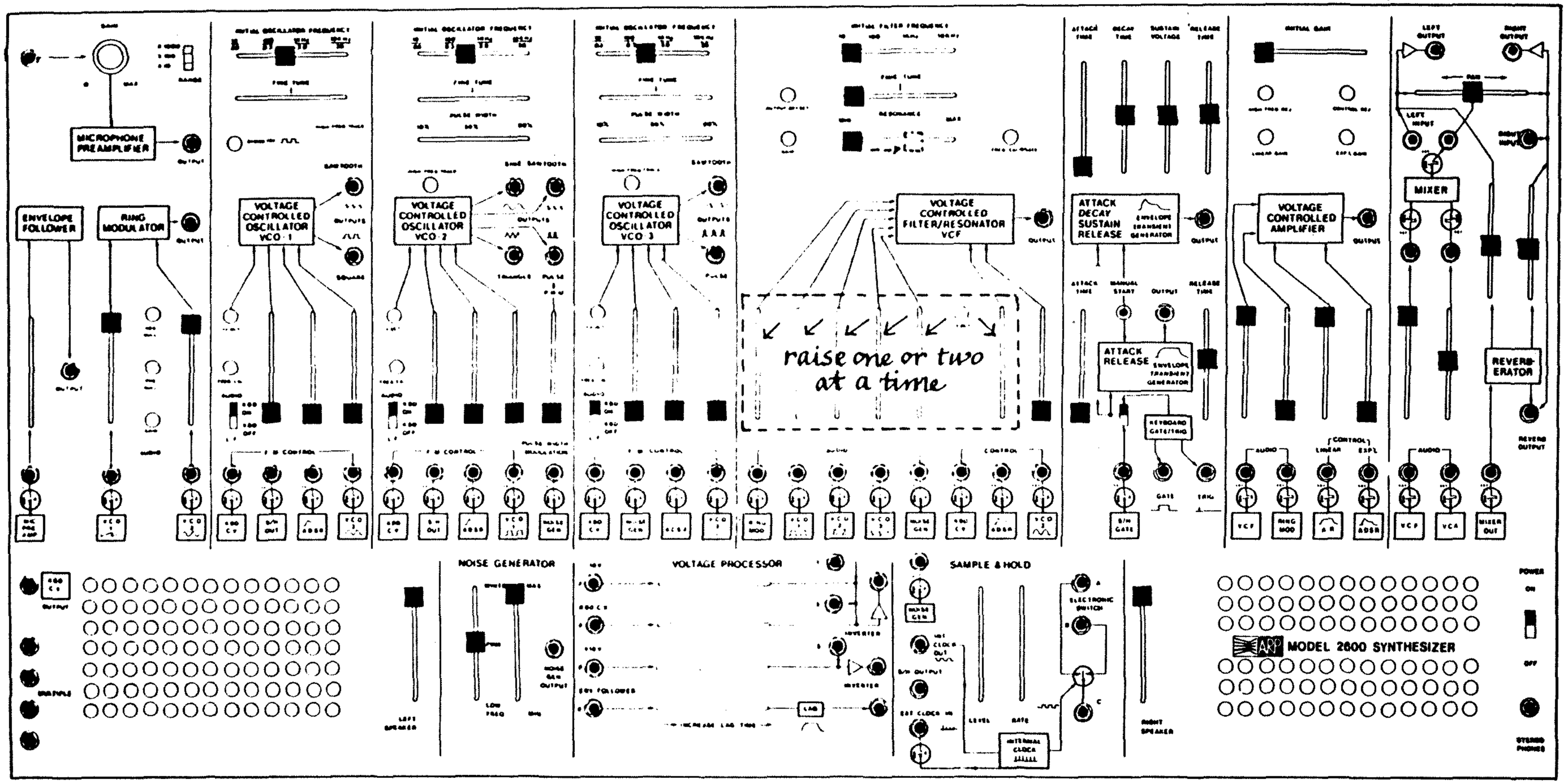
5.13 RM-VCF-VCA

Up to this point you have not used the Ring Modulator. We insert this heading, under which the Ring Modulator is used only with AC coupling and is fed with two oscillators at audio frequencies, to provide you with an excuse for investigating this traditional function, and a place to make notes on the main general results of ring modulation. Begin with the two prewired input signals from the VCO1 sawtooth and the VCO2 sine, and let the signal from the modulator pass unfiltered directly through the VCF to the Mixer. (Note that our prewired connections would also allow you to bypass the VCF entirely by utilizing the prewired RM-VCA input and then opening the VCA-Mixer input rather than the VCF-Mixer input.)

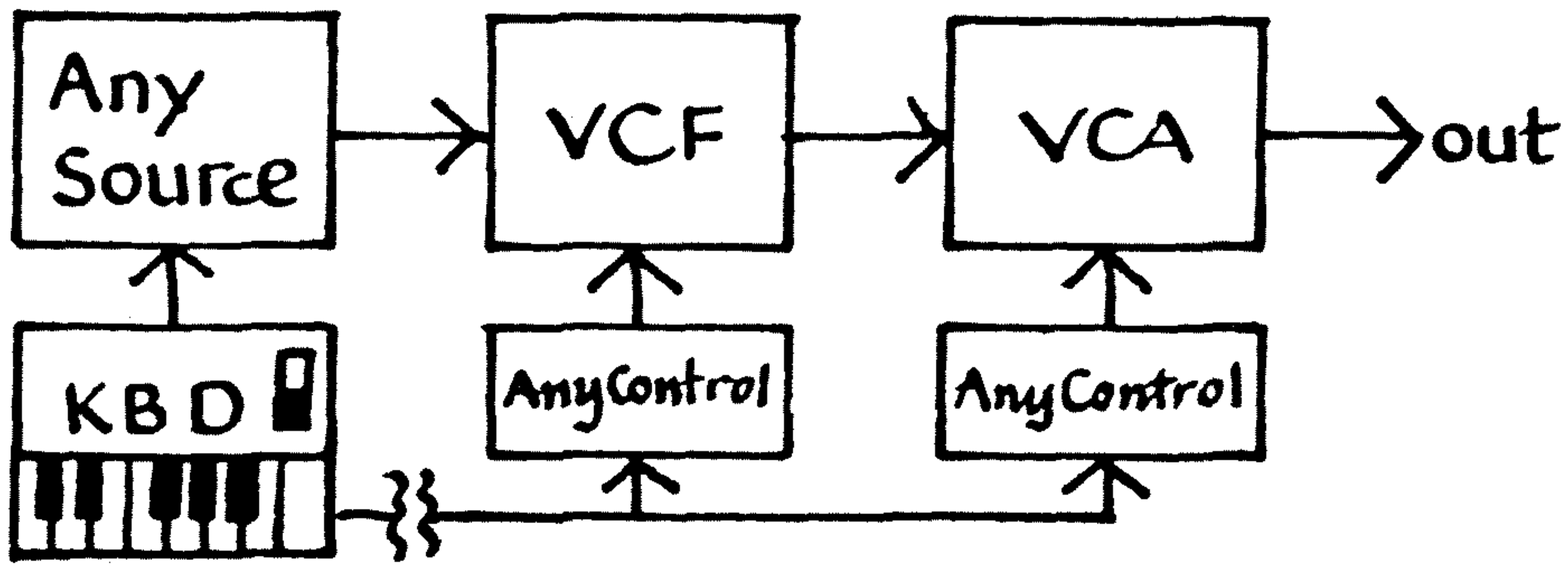


Note what happens when one input frequency is swept slowly while the other remains constant, and also that, whatever character the output from the modulator takes, it remains constant when both oscillators are played from the keyboard. In other words, it should be clear that only the interval between two input frequencies has any effect on the exact timbral characteristics of the RM output; if the two input frequencies move by equal musical intervals, so will the modulator output.

This makes it possible to use the modulator as a means of achieving new timbres that could not be produced from the VCO's and VCF alone.



THE GENERAL PERFORMANCE PATCH

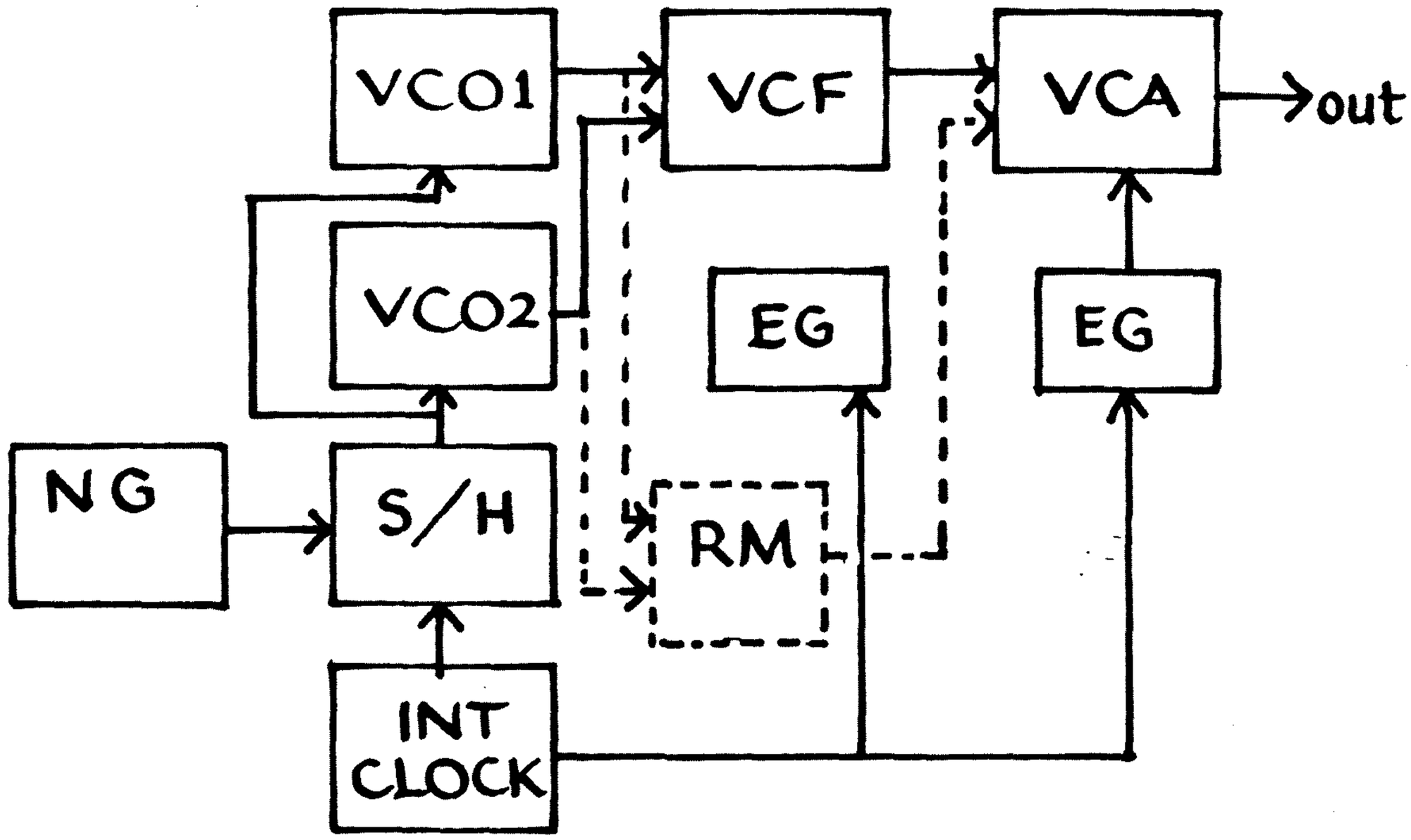


5.14 You've arrived. With this general performance patch we formally give you permission (as if you needed it) to make full, free, and complete use of the keyboard, entirely at your own discretion: if your main and primary interest is live performance using a more or less standard 12-notes-to-the-octave scale, then you already know 90% of what you need to in order to play from now until doomsday without ever repeating the same patch and control settings.

With this patch we also announce that the text of your 2600 instruction manual is coming to a close. You will hear from us from time to time in the patch notes following, and in the alignment ritual given in section 6; but essentially our task is completed.

A few reminders: 1) MAKE NOTES on any particular effect you want to be able to recreate. Even now, when you're learning. Especially now when you're learning; in your later experiments, you'll know exactly what's going on and will be able to remember how particular effects were generated. But right now it's doubtful that you could. Hardly anyone has ever worked with a synthesizer without discovering, very early in his acquaintance with it, at least a few sounds that he never is able to duplicate afterwards.
2) ALWAYS take the time, when you discover a new sound or new effect, to think out exactly how it is being produced and exactly how each element and setting in the patch contributes to the result. One reason for doing this is that occasionally you may find that parts of your patch are doing nothing at all, and you can simplify it accordingly. Another reason is that this is the only way you can emancipate yourself from block diagrams and patch plans. They are, of course, helpful. If they weren't we wouldn't have put such emphasis on using them. But if they multiply beyond all reason and all order, they can become more of a hindrance than a help; you don't want to be forced to rummage through a hundred notebooks every time you turn on the 2600.

5.1401 An alternative form of 5.14, in which two parallel signal paths feed the VCF and VCA respectively before being mixed.



With proper control settings, the 2600 output will be a sequence of notes at random pitches, following one another at a rate determined by the Internal Clock.

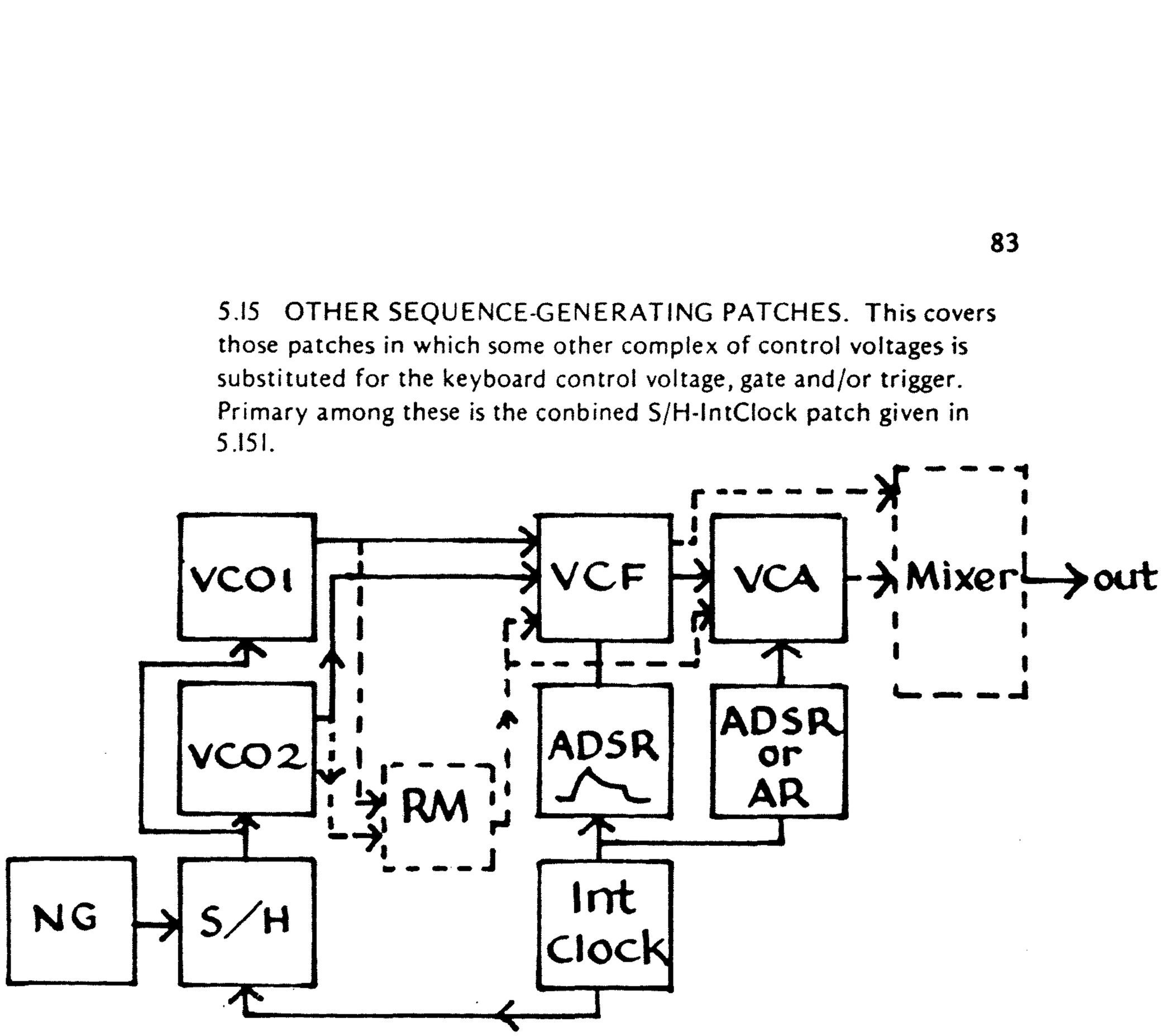
Begin with these control settings and then vary them systematically :

1) VCF and VCA initial frequency and gain at minimum, proper control imputs at maximum.
2) Envelope Generators' gating switch to lower position.
3) VCO's in audio range, about 500Hz; S/H-oscillator inputs about two-thirds open.
4) Internal Clock rate about medium; S/H input level all the way open, and NG output level about three-quarters up, color hardly matters.

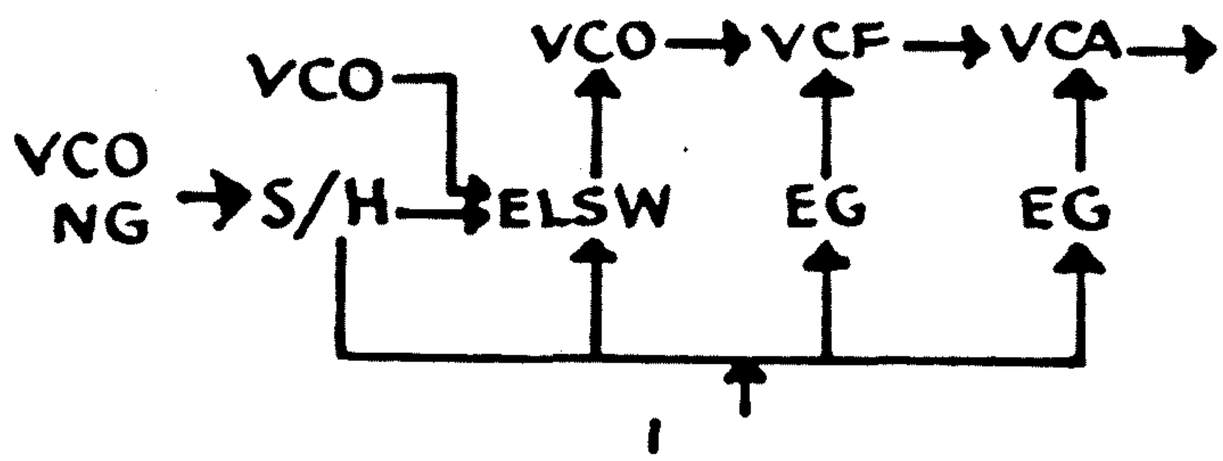
Here are some other possibilities; you may want to diagram them in separate block diagrams.

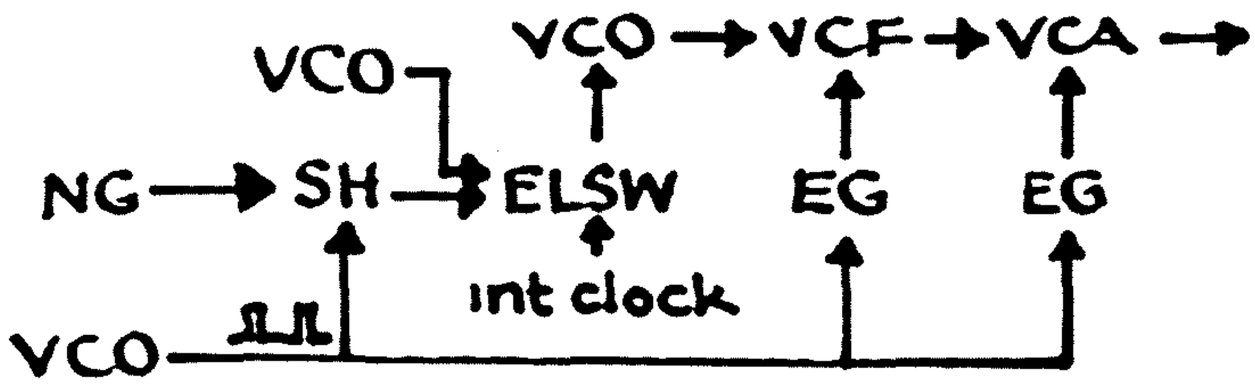
1) Instead of gating the envelope generators from the Internal Clock, switch them back to keyboard gating.
2) Trigger the S/H from the keyboard.
3) Lower the Internal Clock rate to minimum and gate the Envelope Generators from the keyboard. Trigger the S/H from the Internal Clock. Play a tune from the keyboard; sooner or later a changed output voltage from the S/H circuit will change the "key" you are playing in.

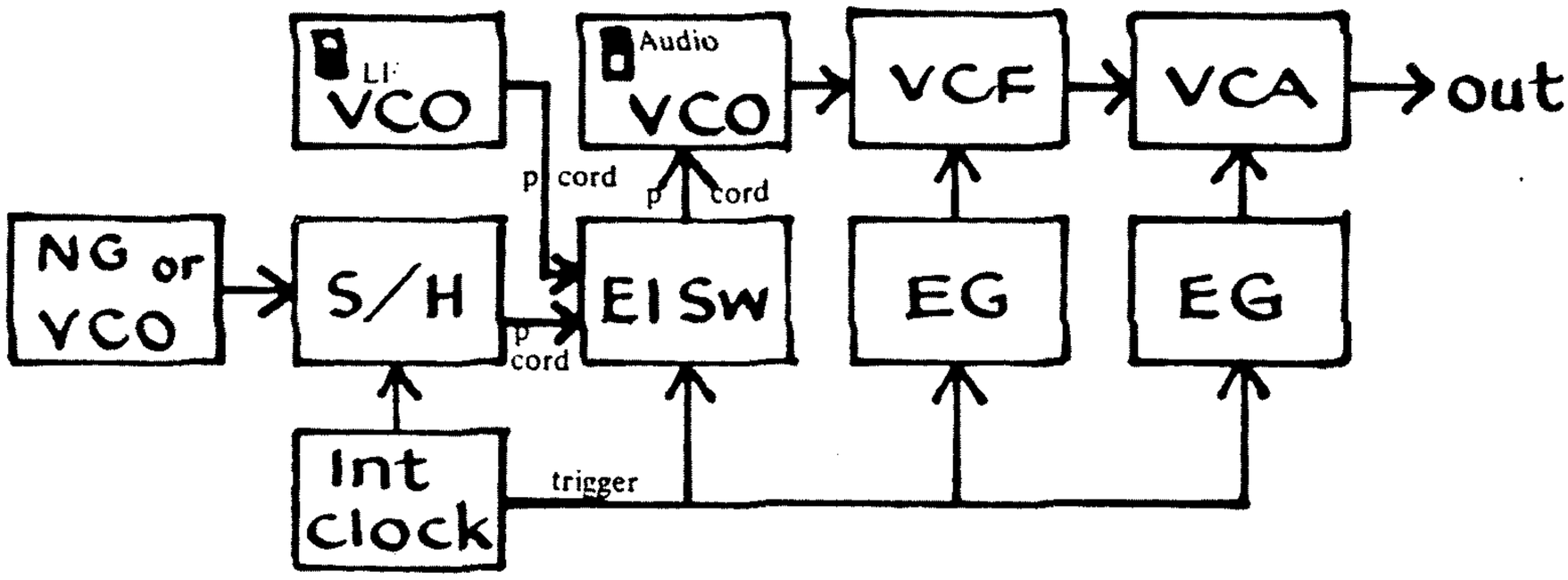
5.15 OTHER SEQUENCE-GENERATING PATCHES. This covers those patches in which some other complex of control voltages is substituted for the keyboard control voltage, gate and/or trigger. Primary among these is the conbined S/H-IntClock patch given in 5.151.



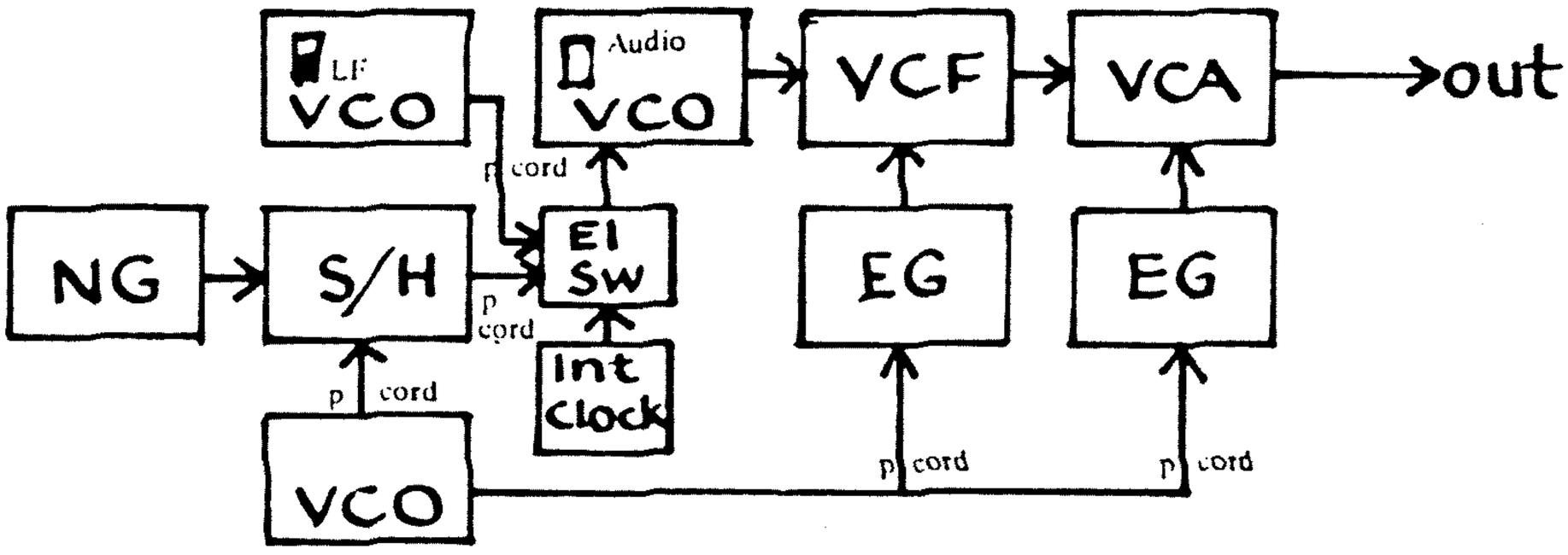
5.1511 Plausible variations on 5.151.





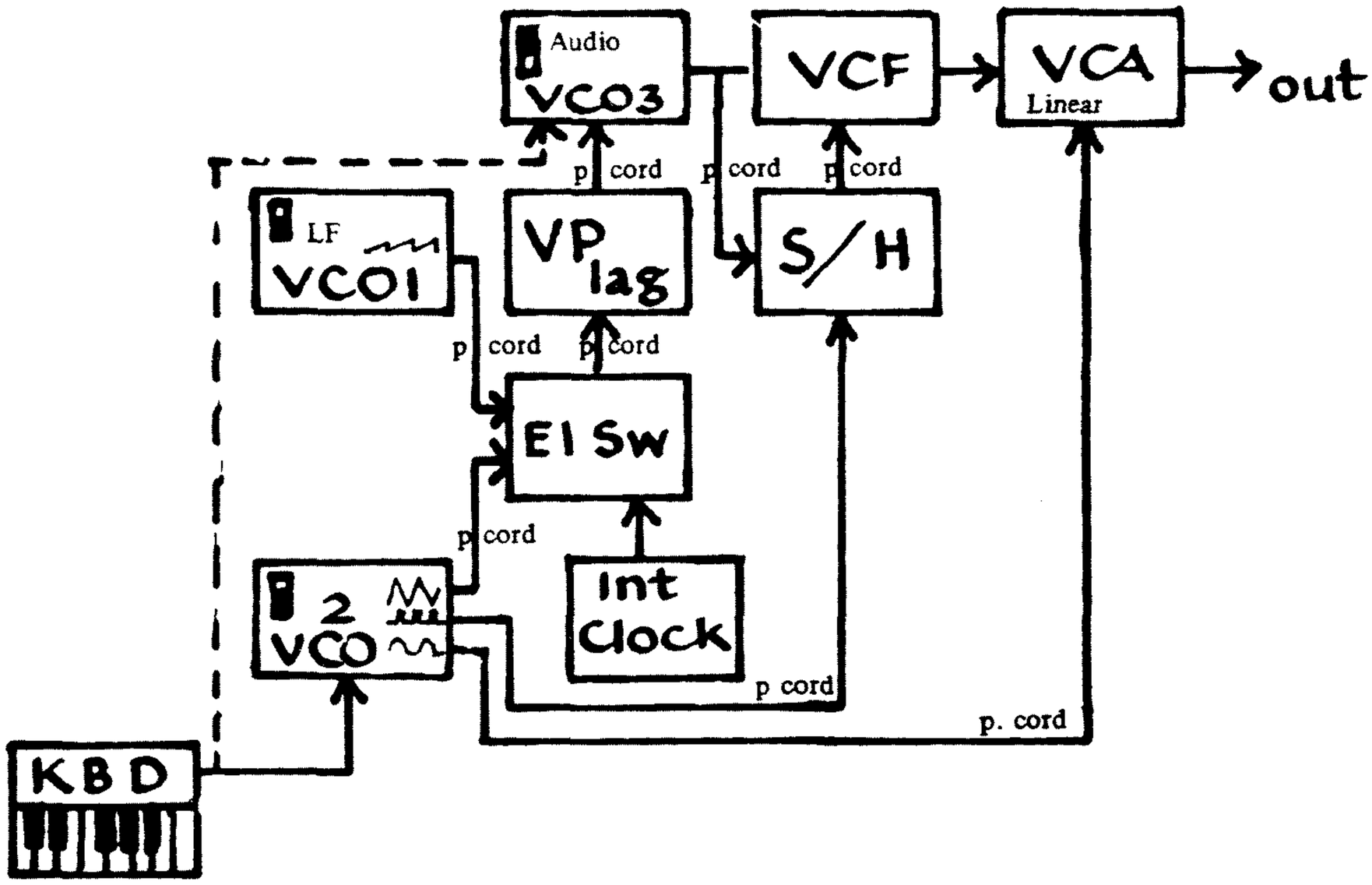


You might, if you wish, begin with a slightly simplified patch ; omit the S/H circuit and feed two VCO's directly into the Electronic Switch. Then experiment thoroughly and systematically. Go on to the patch above. Don't neglect the possibilities of audio-range switching rates; the Internal Clock is capable of rates up to about 100Hz.



Perhaps more useful than the preceding patch. Here the gating of the Envelope Generators is independent of the Internal Clock rate for greater flexibility. Experiment primarily with the three controlling oscillators; the two low-range VCO's and the Internal Clock.

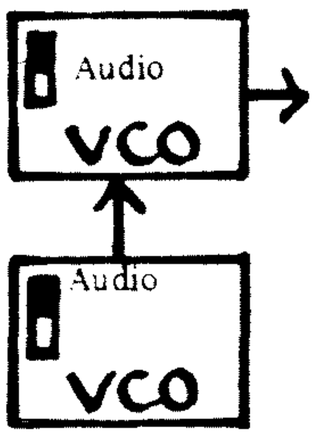
5.152 Another sequential patch. Use the rest of this section to file "complete" patches; i.e. those which produce an audible output more or less continuously without requiring interference from you. In many of them, of course, use of the keyboard is at least optional; but it should not be necessary. (The patches you will file under 5.14 will be those which produce no output unless you play them from the keyboard).



The dotted-line control path is prewired ;omit it at first by inserting a dummy plug into the KBDCV-VCO3 input. Adjust the VCA initial gain and linear control input for intermittent sound from the sine-wave peaks. The keyboard will govern only the frequency of these events and not the pitch of VCO3. Both of the controlling oscillators would be set for periods not exceeding lHz at first; suitable changes in the frequency of the Internal Clock and the setting of the Lag Processor will produce changing patterns of control voltage for the audio oscillator VCO3.

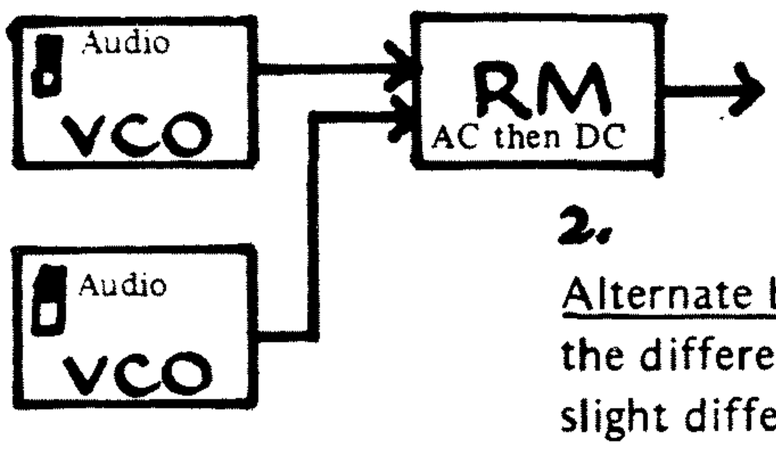
5.2 INCOMPLETE PATCHES INVOLVING INDIVIDUAL FUNCTIONS.

These form parts of complete patches. In the first class are patches whose output is primarily at audio frequencies and is intended to provide a signal for further processing and 'packaging' into discrete events. In the second class are those patches whose output is primarily at low frequencies and is intended to provide interesting varieties of control over a separate signal path.



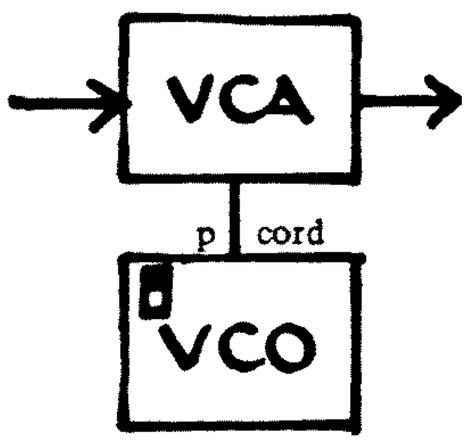
1.

Frequency Modulation. Note the effects of varying the depth of modulation, and the interaction between the control frequency and modulation depth. Use this section to catalogue those effects you find useful.



2.

Alternate between AC and DC coupling for identical signals and note the difference in the sound. The Ring Modulator enormously exaggerates slight differences in frequency ; just try to zero-beat the output of two oscillators by listening to them through the RM. Use this section to catalogue settings you find useful.

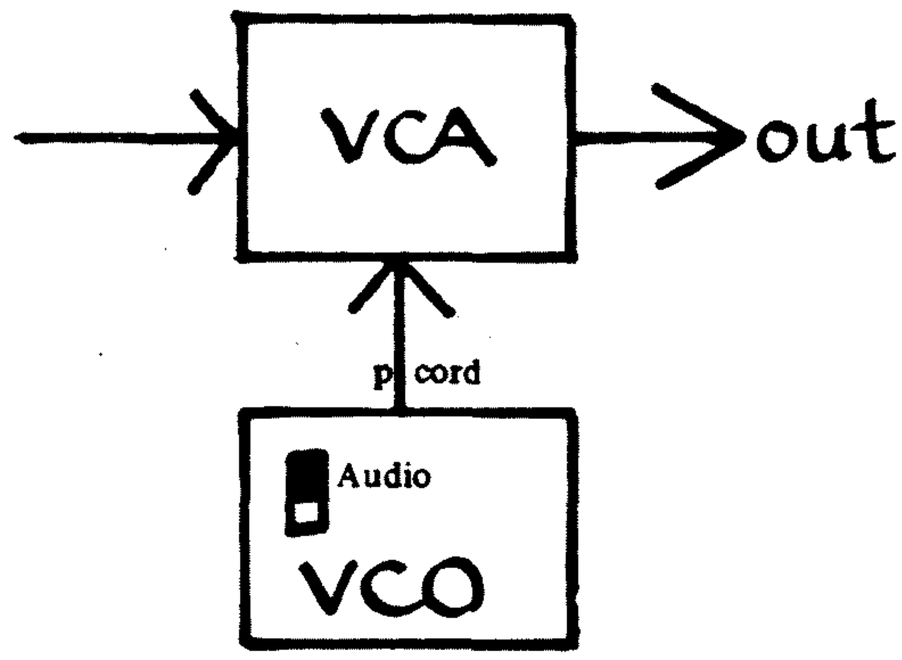


3.

Call tnis 'timbral modulation' or 'harmonic modulation' or anything you please. There is no word for it. Try it first with a noise input to the VCF and a sharply resonant Q-setting. Be sure the VCO is in audio range and then open slowly the control input to the VCF. The VCO2 sine is prewired don't neglect others, and even combinations of different waveforms from different oscillators. Use this section to catalogue settings you find useful. And experiment with control by more than one oscillator at once.

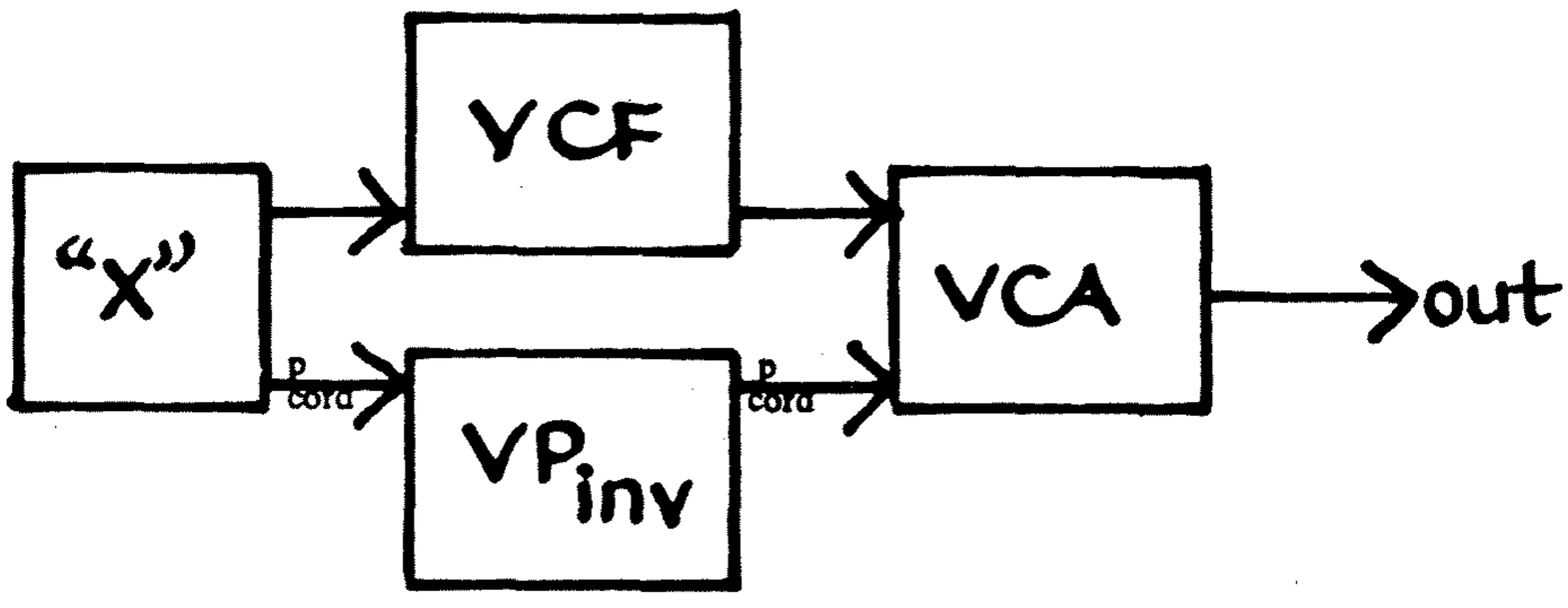
4.

Note that like the preceding two patches this one covers only audio-frequency modulation—in this case, amplitude modulation. Most of these you are somewhat familiar with already. Use this section to catalogue effects and settings you find useful. And experiment with control by more than one oscillator at once.



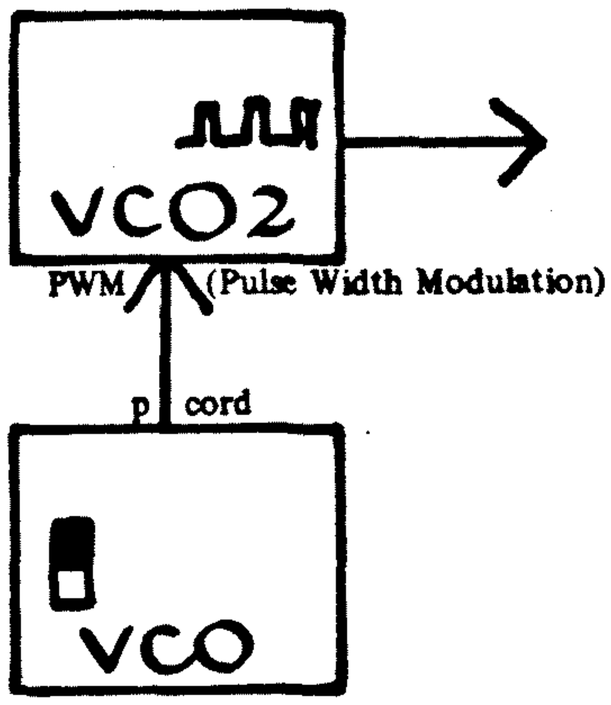
5.

Inverting any signal and combining it with a filtered version of itself will result in cancellation of the unfiltered components of the signal. *In other words, this is* how to turn a low-pass into a high-pass filter. Level settings are quite critical--tune them to a null at the VCA output while the VCF initial gain is all the way up and Q is at minimum. Use the Noise Generator to begin with, and patch it to one of the unattenuated inverter inputs.



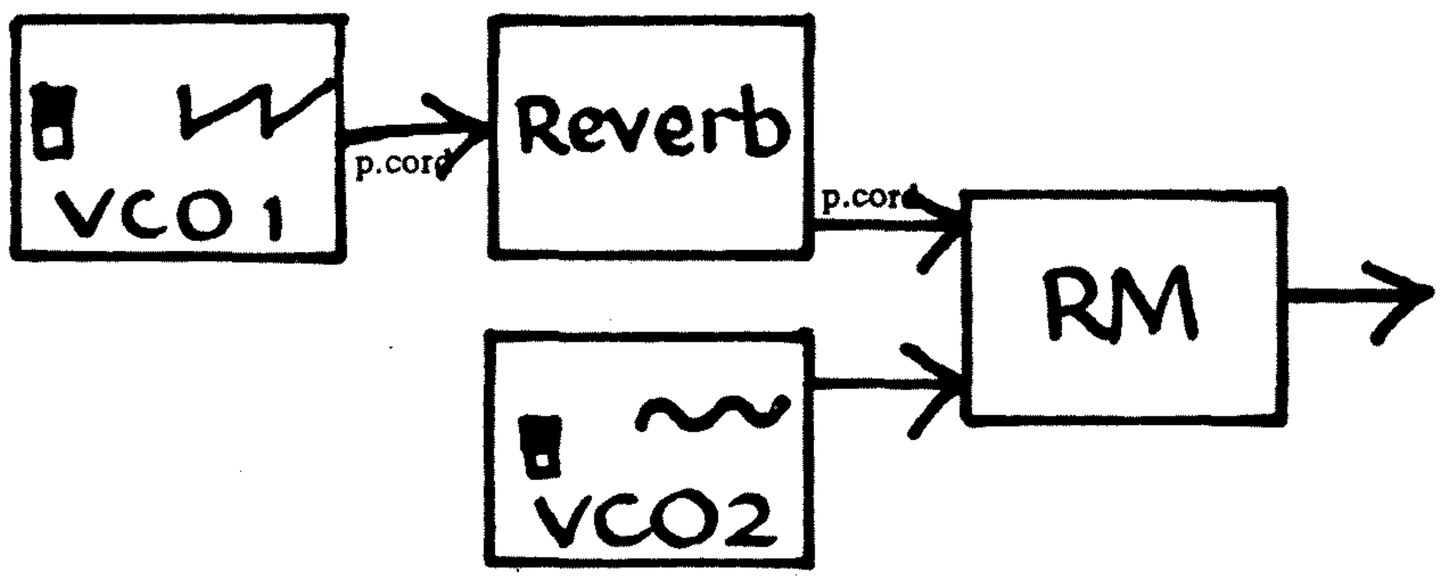
6

Pulse-width modulation. Since only the trailing edge of the pulse is affected, AF modulation of the pulse will cause the trailing edge to become a series of shorter pulses at a frequency equal to the modulating frequency. A number of timbral effects becomes possible in this fashion that would be difficult to achieve by any other means. Note that if the modulating signal is a positive-going waveform only, such as the 2600's square, pulse, and sawtooth waves, then the initial setting of the pulse width control should be at minimum; but if the modulating signal is balanced on either side of OV then the initial pulse-width setting should be at 50%.



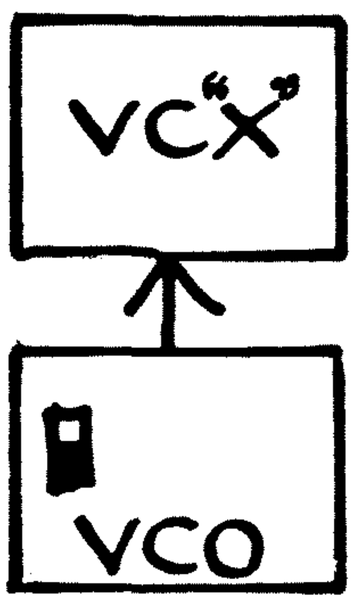
7.

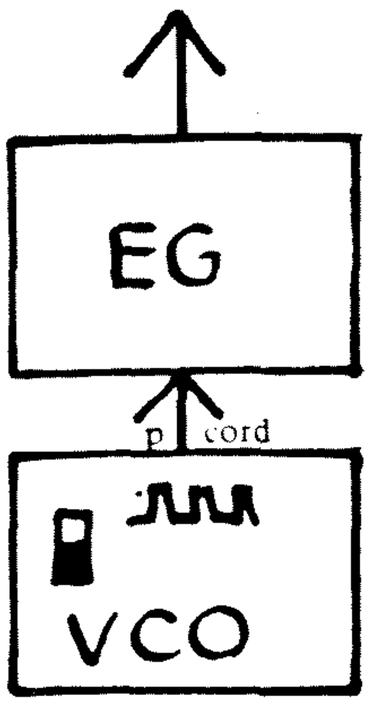
Whenever you need a reverberated signal not for listening but for further processing, take your signal from the REV out jack just under the panel box indicating the reverb unit. This will automatically disconnect the right-channel reverb output from the final output stages of the 2600. Leave the left-channel reverb output level control all the way down for this particular patch, and feed VCO1 directly into the reverb input with a patch cord. In situations such as this one the reverb unit can improve enormously the gong-like imitations you can get from the RM. But this is a little-explored field and you are on your own here.



8.

Of course, you are familiar with this. Use this section, however, to catalogue particular settings you find useful, and to experiment with multiple waveforms and control sources.



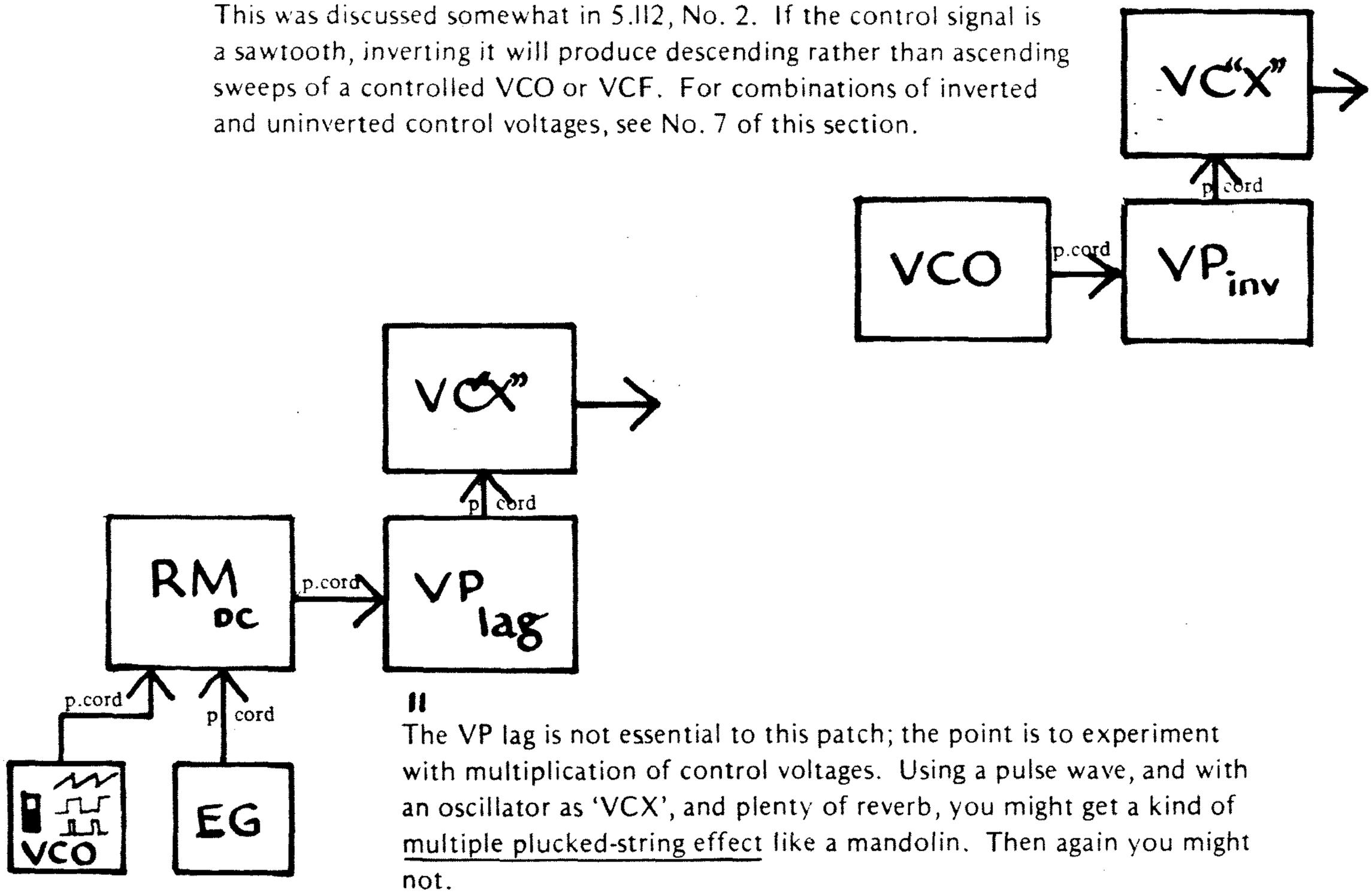


9

You should get used to the idea that there can be other gate sources for the envelope generators besides the keyboard and the Internal Clock. Any square or pulse wave--in fact any steep-rising wavefront--can function as a gate, and/or trigger. Use the above patch, for example, whenever you require voltage control of the repetition--rate of the Envelope Generators The Internal Clock is not subject to voltage control. If you use VCO2 in a situation like this, you can have some control also over the duration of envelopes by modulating the pulse width of the VCO2.

10

This was discussed somewhat in 5.112, No. 2. If the control signal is a sawtooth, inverting it will produce descending rather than ascending sweeps of a controlled VCO or VCF. For combinations of inverted and uninverted control voltages, see No. 7 of this section.

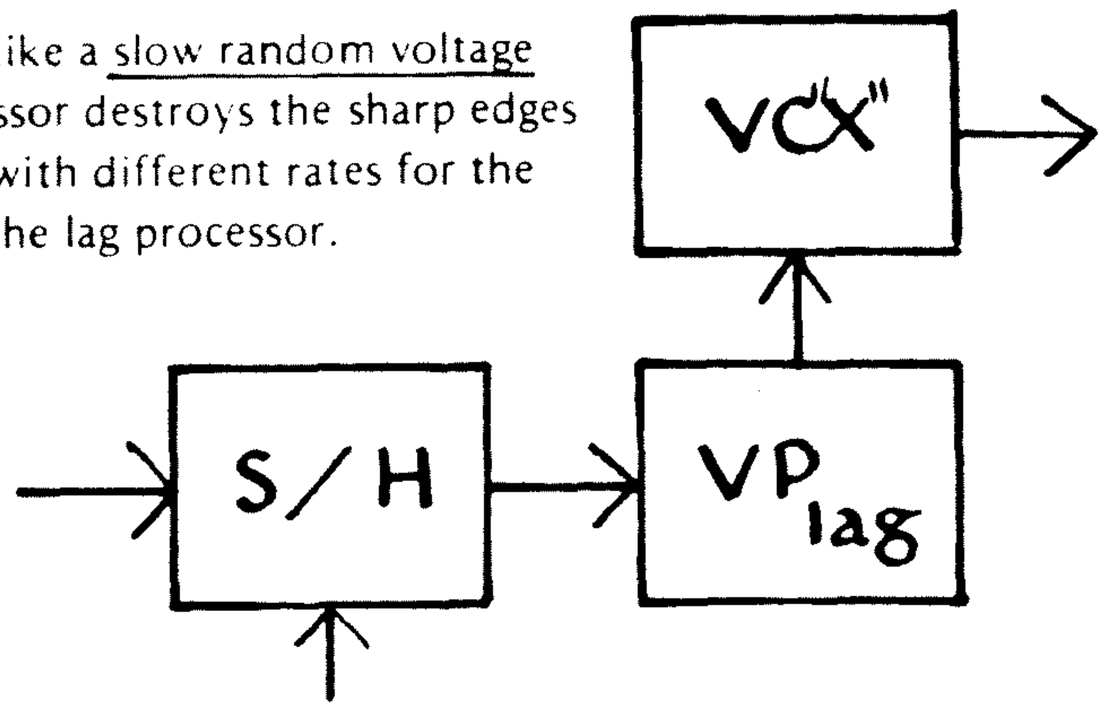


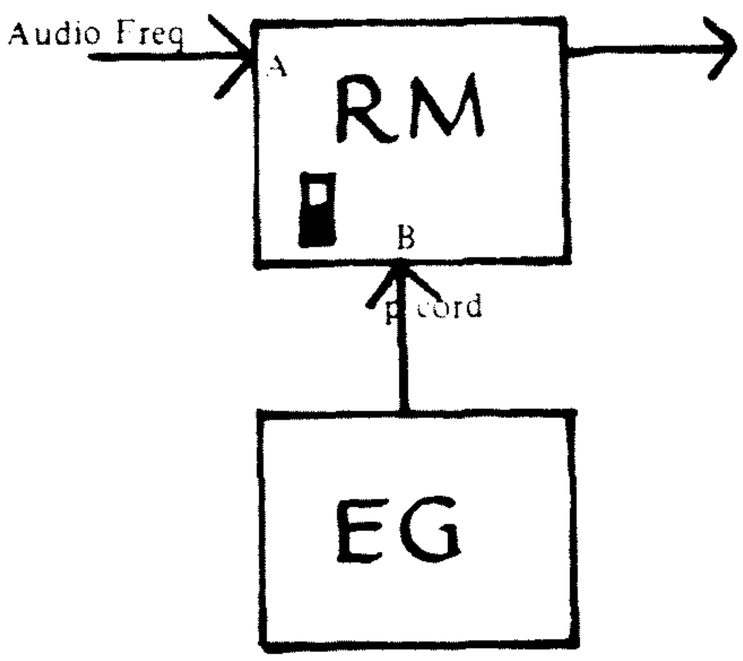
11

The VP lag is not essential to this patch; the point is to experiment with multiplication of control voltages. Using a pulse wave, and with an oscillator as 'VCX', and plenty of reverb, you might get a kind of multiple plucked-string effect like a mandolin. Then again you might not.

12.

The point of this is to create something like a slow random voltage to control anything with. The lag processor destroys the sharp edges or steps in the S/H output; experiment with different rates for the Internal Clock and different settings of the lag processor.





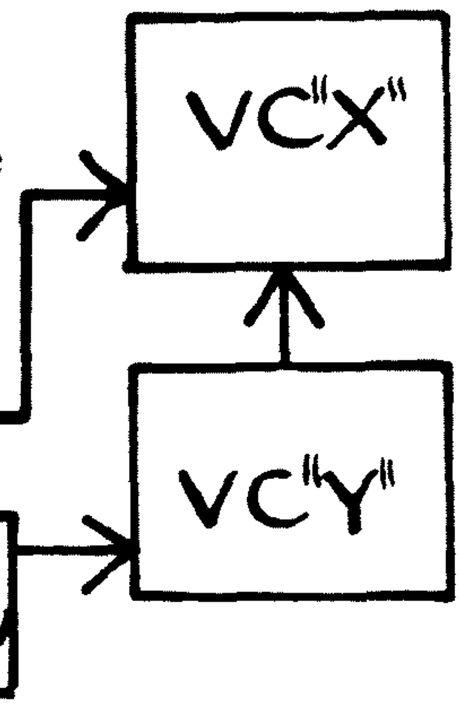
13.

The basic patch for using the Ring Modulator as a VCA. Note that, unlike a true VCA, the Ring Modulator will produce an output from any input voltage whatsoever, regardless of whether it is positive or negative. For example, patch in a voltage inverter between the EG and the modulator; the output will remain the same as it was except that it will be reversed in phase as if it itself had been passed through the inverter.

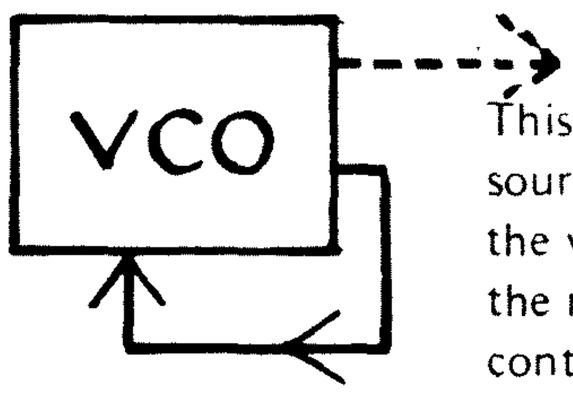
14.

The basic patch for producing opposite effects from the same control voltage. VCX takes a direct signal, and VCY takes an inverted signal. In conjunction with No. 6--using the RM as a VCA--this gives rise to some interesting automatic-pan possibilities. Let VCX be the Ring Modulator and VCY by the VCA with initial gain at maximum; for the CV source use a slow sawtooth, and feed an identical audio signal into both the RM and the VCA. Feed the two outputs to opposite channels, and the result will be a slow automatic pan from one side to the other, with a 'snap-back' to the original channel. See also some of the external equipment patches in 5.5I.

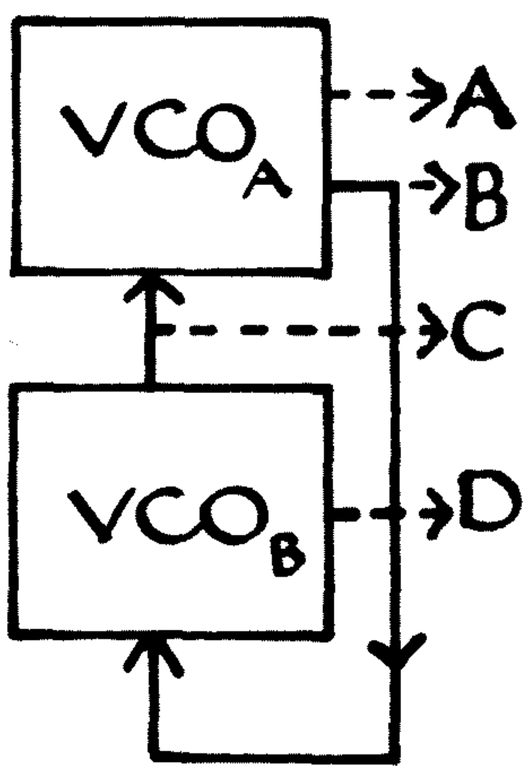
If VCX and VCY are oscillators, and the keyboard is the voltage source, one VCO will descend in pitch as the other rises.



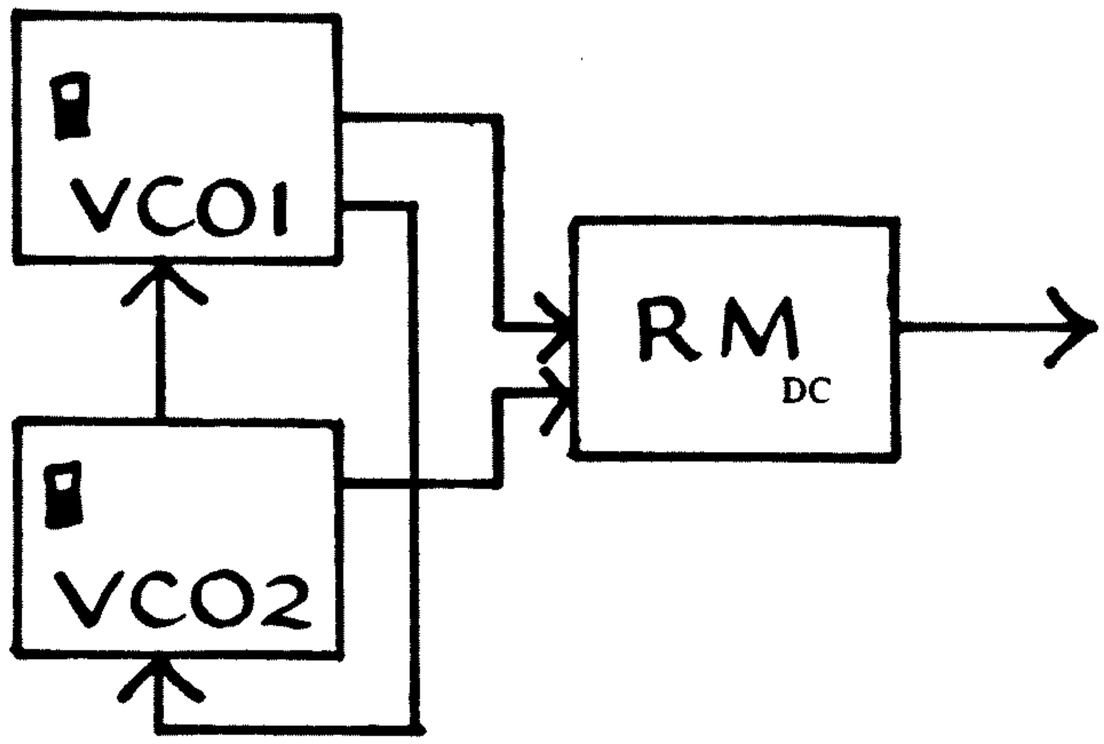
5.3 FEEDBACK LOOPS. A little-explored no-man's-land, including control of an oscillator by its own output voltage, mutual control of two oscillators each by the others output voltage, and other horrors.



This kind of feedback appears to be most usefull as a control-voltage source, not so much an audio-signal source. But try it both ways. When the waveform being fed back is one of the positive-going forms, increasing the modulation depth will drive the VCO up in pitch, and some further control voltage must be used to counteract this--some negative DC.
For the same reason, a VCO subjected to feedback of this sort will not respond to keyboard control with equal musical intervals. This is one reason why feedback is not terribly useful as a means of generating audio signals.



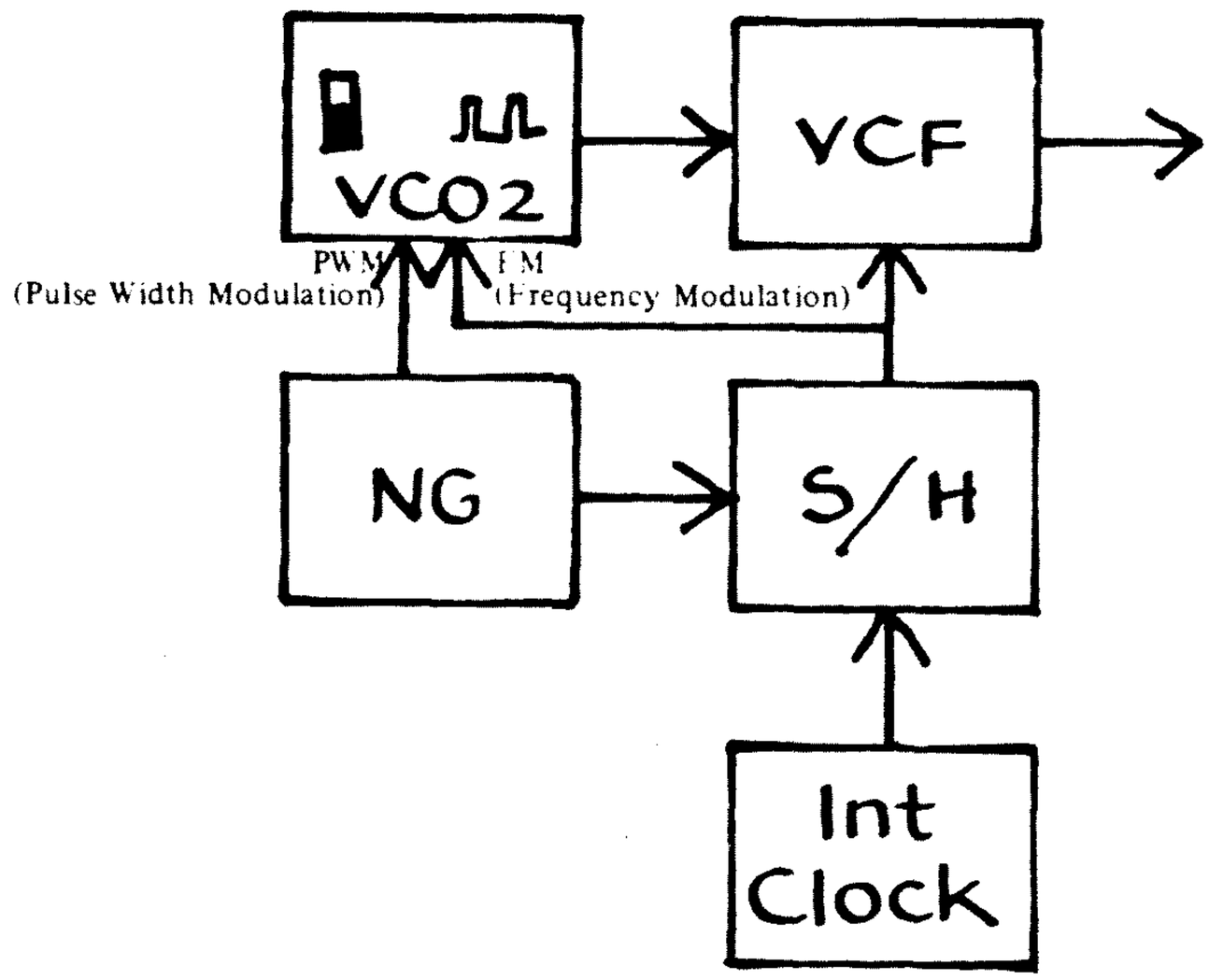
2.
The four dotted lines indicate possible control-voltage taps. A and D are taps from other waveforms than those which are being fed from VCOA and VCOB and vice versa; B and C tap those voltages directly--for, of course, the shape of the waveform at those outputs is affected by the mutual feedback just as much as the shape of any other waveform available from either of the two oscillators.
In general, the modulation depth settings for feedback are quite critical of the result is to be 'interesting'. One VCO will almost completely override the other one unless the relative depth of modulation for the two control-voltage inputs is carefully balanced.



4.

Essentially this is a refinement of No. 2 in which the two VCO outputs are multiplied by each other before finding their way to some control input. Note that this multiplication can be combined with the use of one or more of the VCO outputs directly, as indicated by the dotted lines.

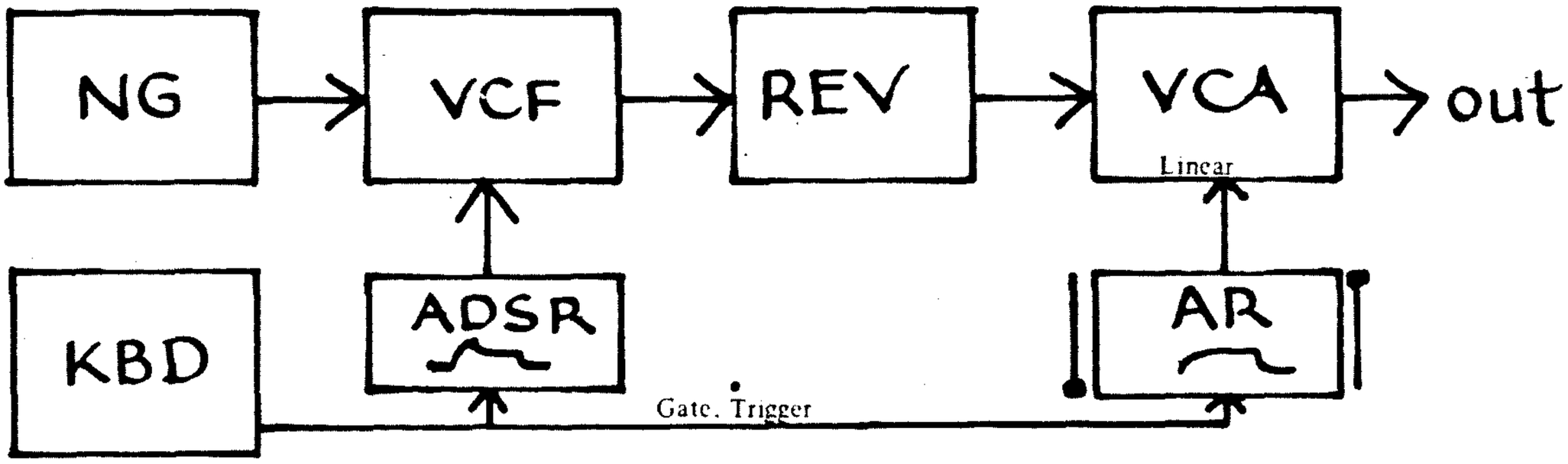
5.4 SOUND EFFECTS. Here is a place to put patches that specifically imitate natural sounds. If you wish to put musical-instrument sounds in here, you might make a separate sub-classification.



1.

In general, control settings for any imitations of natural sounds will be quite critical. In this case, begin with the VCF wide open and tune the VCO to an initial frequency and pulse-width setting that catches what feels like the right tempo for footsteps. Then retune this by adding a small amount of noise modulation to the pulse-width, closing the VCF perhaps a little way, adjusting perhaps the color of the noise, and adding at a low stepping rate a little FM from the S/H circuit. This last step is a kind of super-refinement; it produces slight irregularities in the speed of the footsteps. If, finally, you add a little of the S/H output into the VCF control, you will get the effect of walking over different kinds of material or in different acoustic surroundings as the S/H voltage changes, by small amounts, the Fc of the filter.

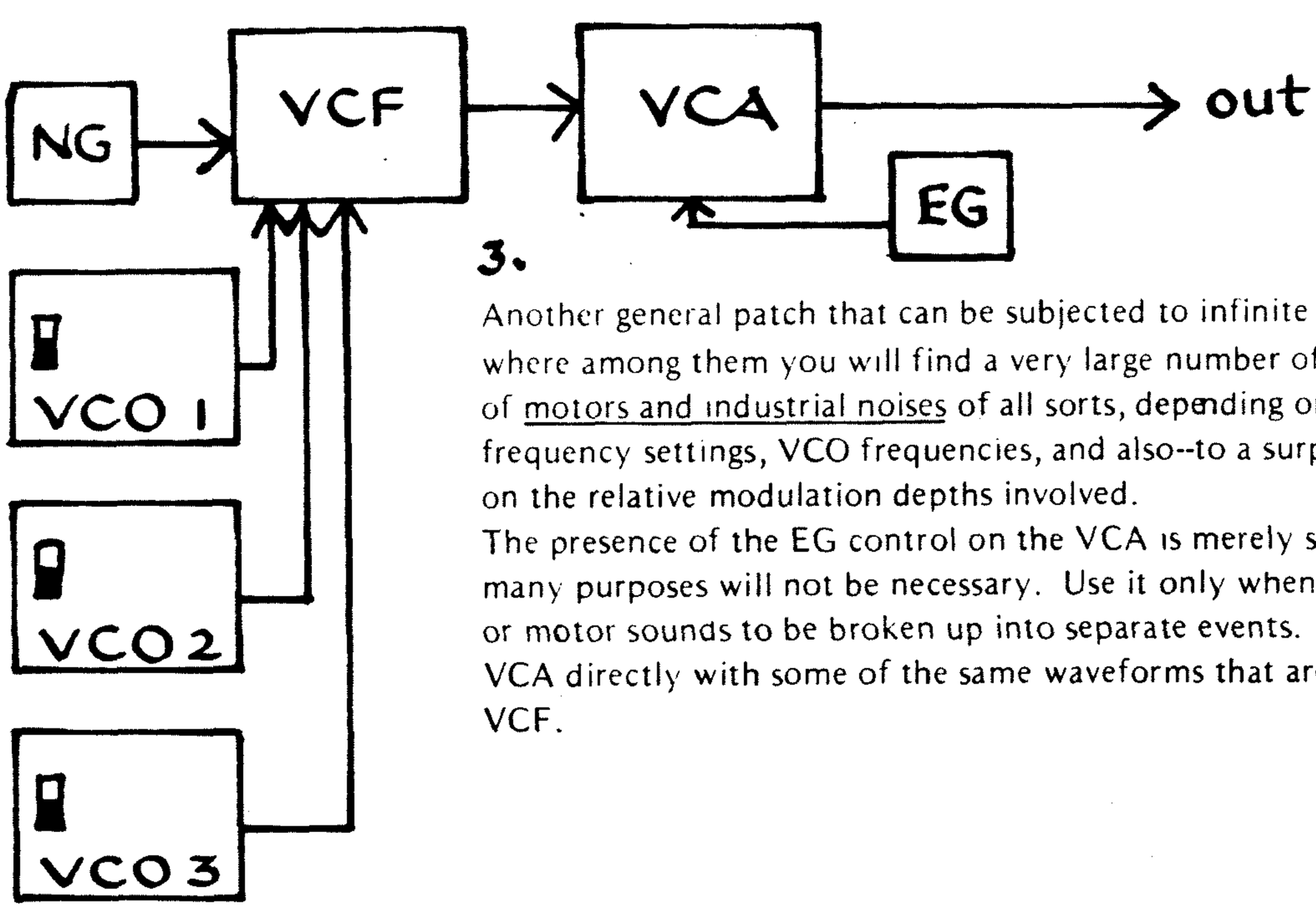
Thunder!



2.

A general patch that is already basically familiar to you from the early experiments of this chapter. The only refinement is the introduction of the REV unit between the filter and the amplifier. You may prefer the effect with the REV in its standard position in the patch--try it both ways.

Another interesting factor in this patch is the use of the keyboard to control the Envelope Generators for other than melodic purposes.



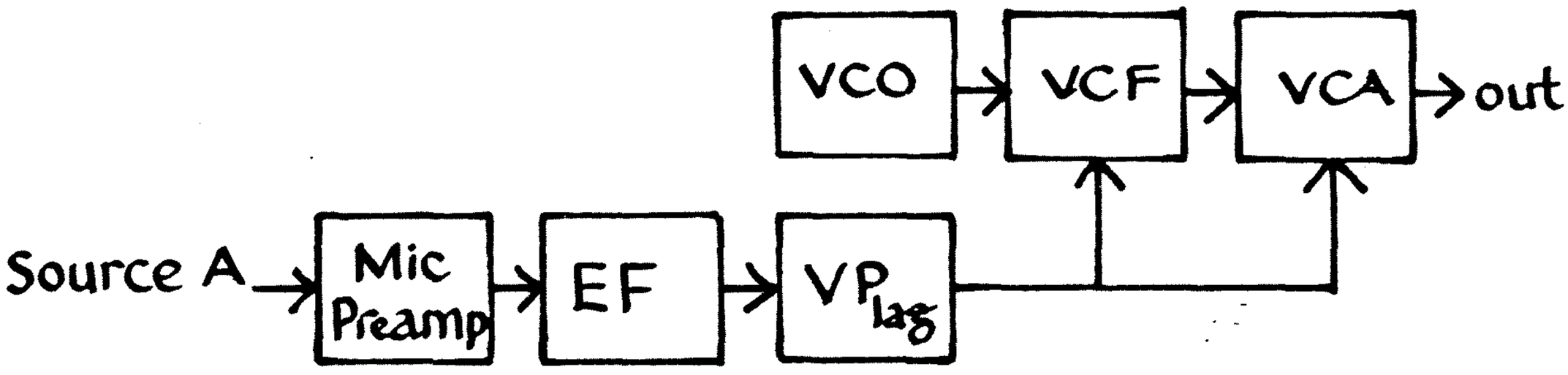
3.

Another general patch that can be subjected to infinite variations. Somewhere among them you will find a very large number of sounds suggestive of motors and industrial noises of all sorts, depending on the initial VCF frequency settings, VCO frequencies, and also--to a surprisingly great degree--on the relative modulation depths involved.

The presence of the EG control on the VCA is merely schematic and for many purposes will not be necessary. Use it only when you require machine or motor sounds to be broken up into separate events. Or control the VCA directly with some of the same waveforms that are controlling the VCF.

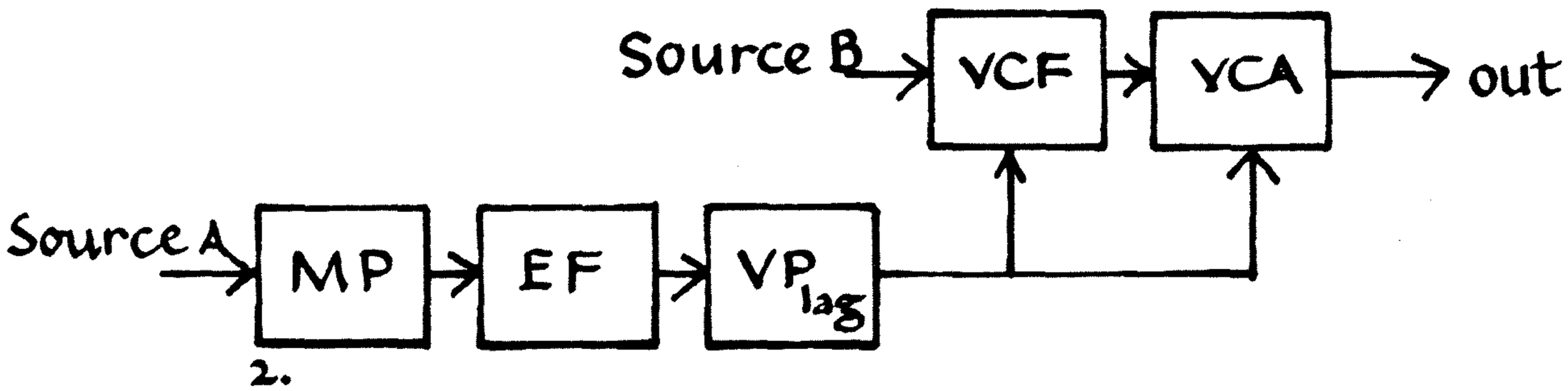
5.5 USING EXTERNAL EQUIPMENT. We give a few general patches here, in two categories. In the first are all those patches in which the 2600 is used to modify externally generated signals from e.g. electric instruments or tape recorders. In the second are all those patches in which external equipment is used to modify or to store signals generated by the 2600.

5.51 Processing externally-generated signals.



1.

The basic patch for using an external source to control some operating function of the 2600. The Microphone Preamp may not always be necessary; but the use of the Envelope Follower is essential. Experiment with the VP lag in and out of the circuit--its only function here is to smooth out the EF output a little.

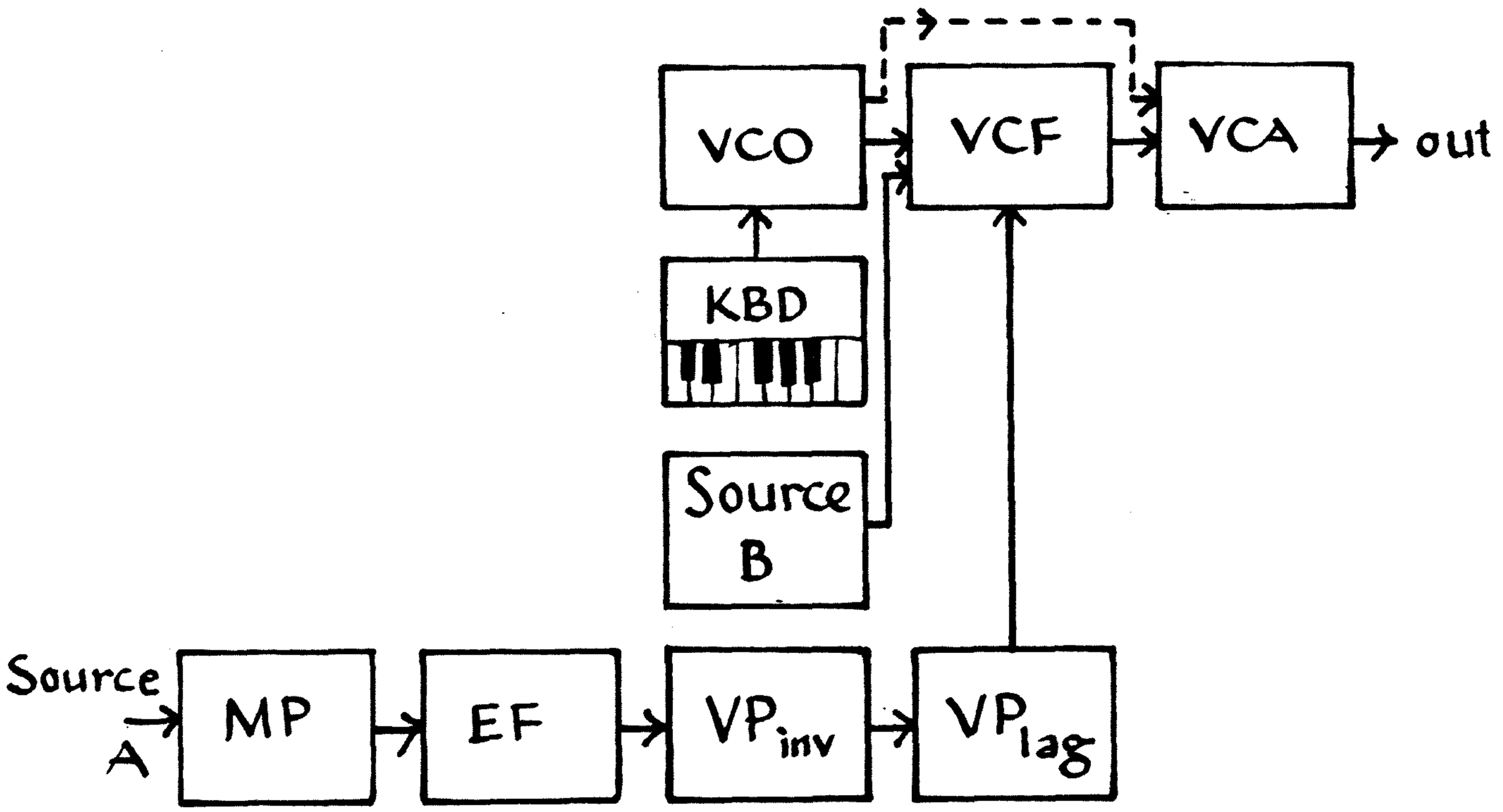


2.

An obvious improvement on No. 1. If, for example, source B is a Hammond, or the vocals of a rock group, and source A is a lead guitar, then the guitar envelope will modulate the organ or vocal sound. There are innumerable possibilities contained even in such a simple patch as this one.

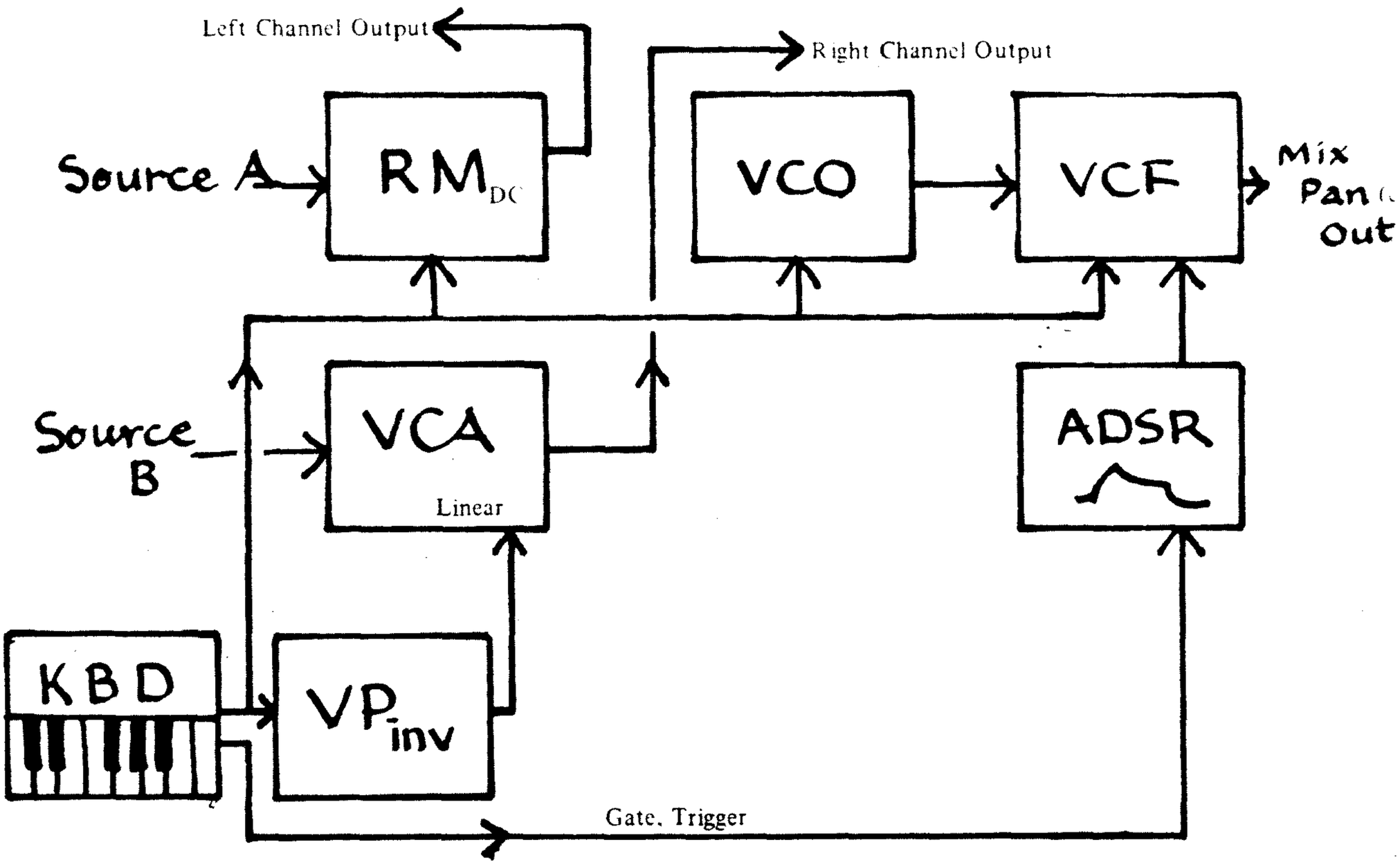
3.

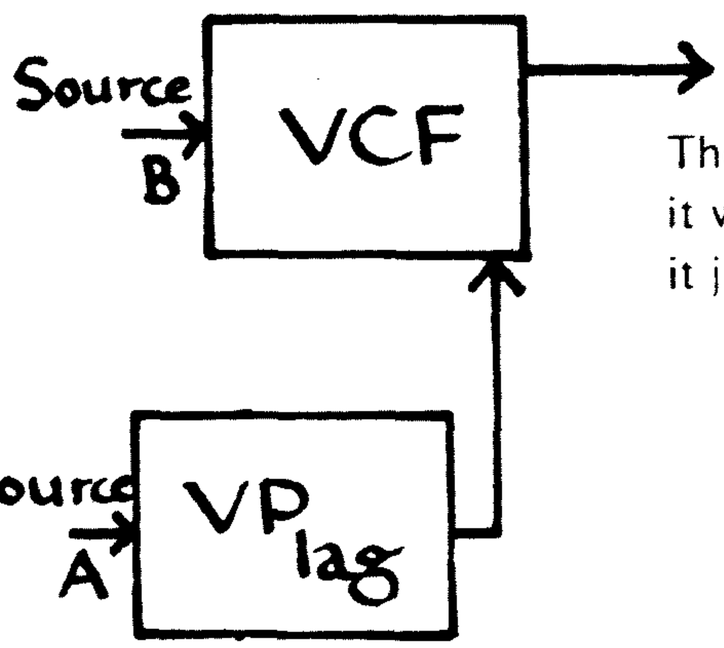
A combination of No. I and No. 2 in which the signal output of the 2600 will consist both of internally generated sounds and external ones (source B) modulated by the envelope of source A.



4

This will require some careful adjustment. The final effect should be that as the melody played on the 2600 keyboard ascends, source A will grow louder, and as the keyboard melody descends, source B will grow louder while source A grows softer. If A and B are the same this amounts to a keyboard-controlled automatic pan. Note that the pan effect is independent of whether the VCO-VCF-Mix-Pan-signal path is actually used or not; it is derived only from the keyboard voltage alone.



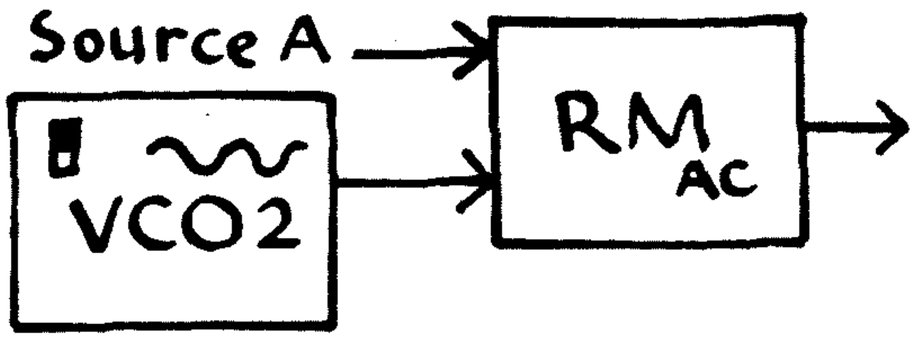


5.

The lag processor does not make a terribly good envelope follower, but it will make some interesting control voltages out of an audio signal. Try it just for the experience.

6.

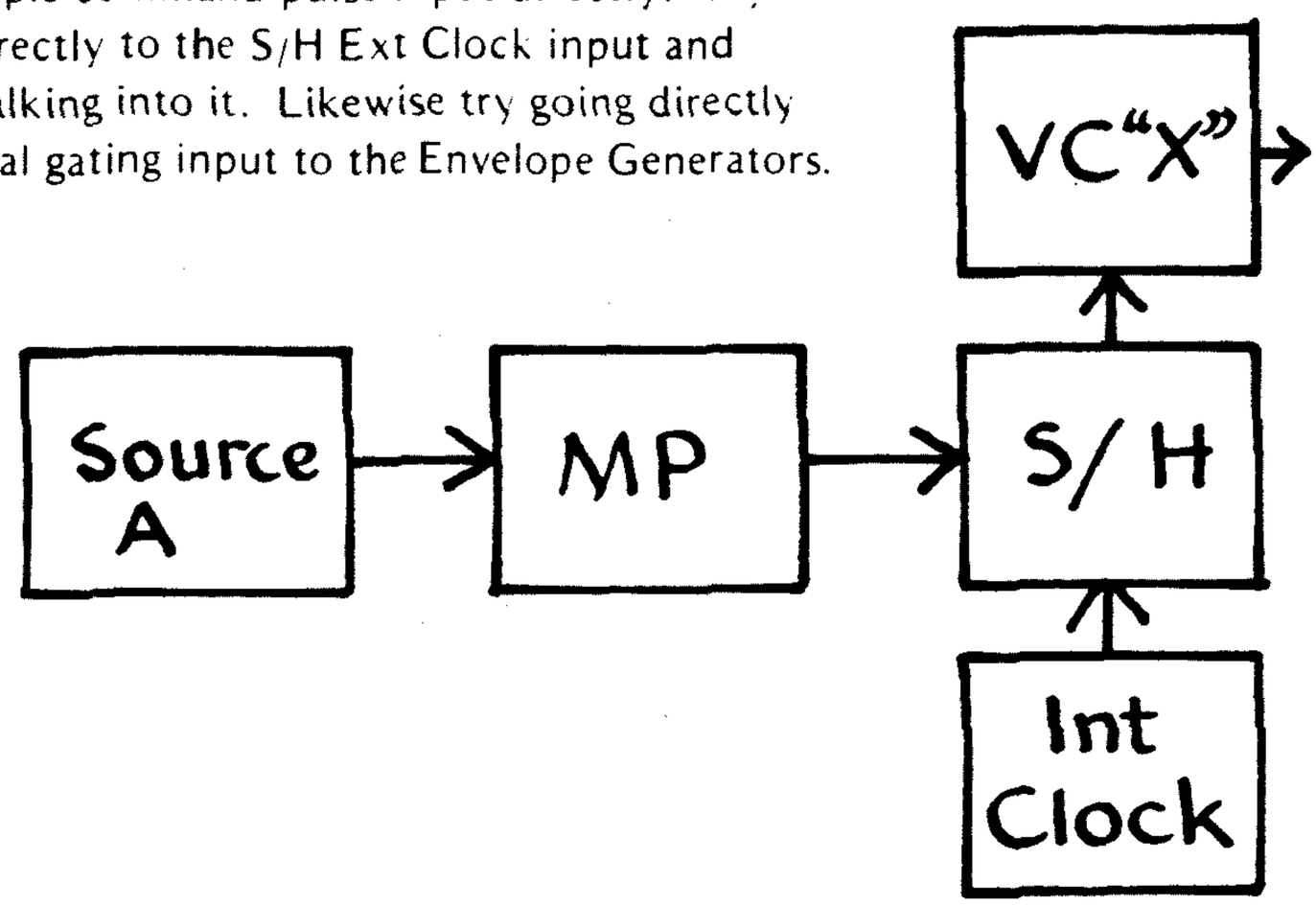
You know what the RM does with a sine and a sawtooth. Now feed it with a sine and a live signal--say from a microphone. Sing into it with the VCO tuned to about 200Hz. Play the VCO from the keyboard while you are singing. Substitute any chord from the flute stops of an electric organ. Substitute the organ for VCO 2. Any relatively pure tone will work well as the second RM input. Note, however, that if both input waveforms are extremely complex, the RM output becomes impossibly so--harsh and scratchy.



7.

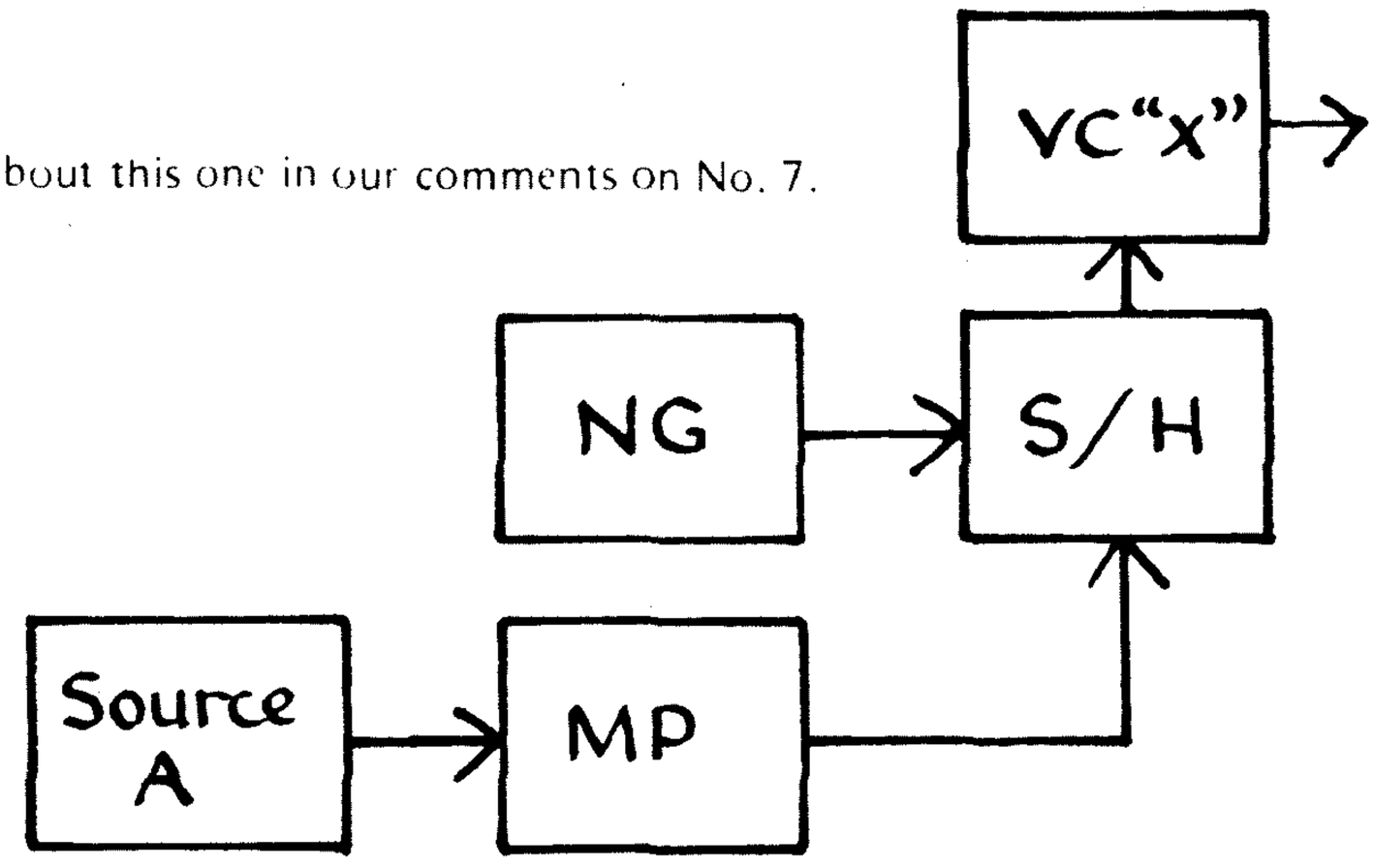
With the patch given here, the S/H output voltage will remain relatively constant so long as no signal comes from source A. If A is a microphone, talk into it, and immediately the S/H output voltage will assume the apparently random sequence of values you are familiar with. The only interesting characteristic of this sampling of an external signal is that this gives you external control over the maximum deviation assumed by the control voltage.

Incidentally, many standard audio waveforms have enough steep-rising wavefronts to trigger the sample command pulse input directly. Try connecting the MP output directly to the S/H Ext Clock input and tapping the microphone or talking into it. Likewise try going directly from the MP into the external gating input to the Envelope Generators.



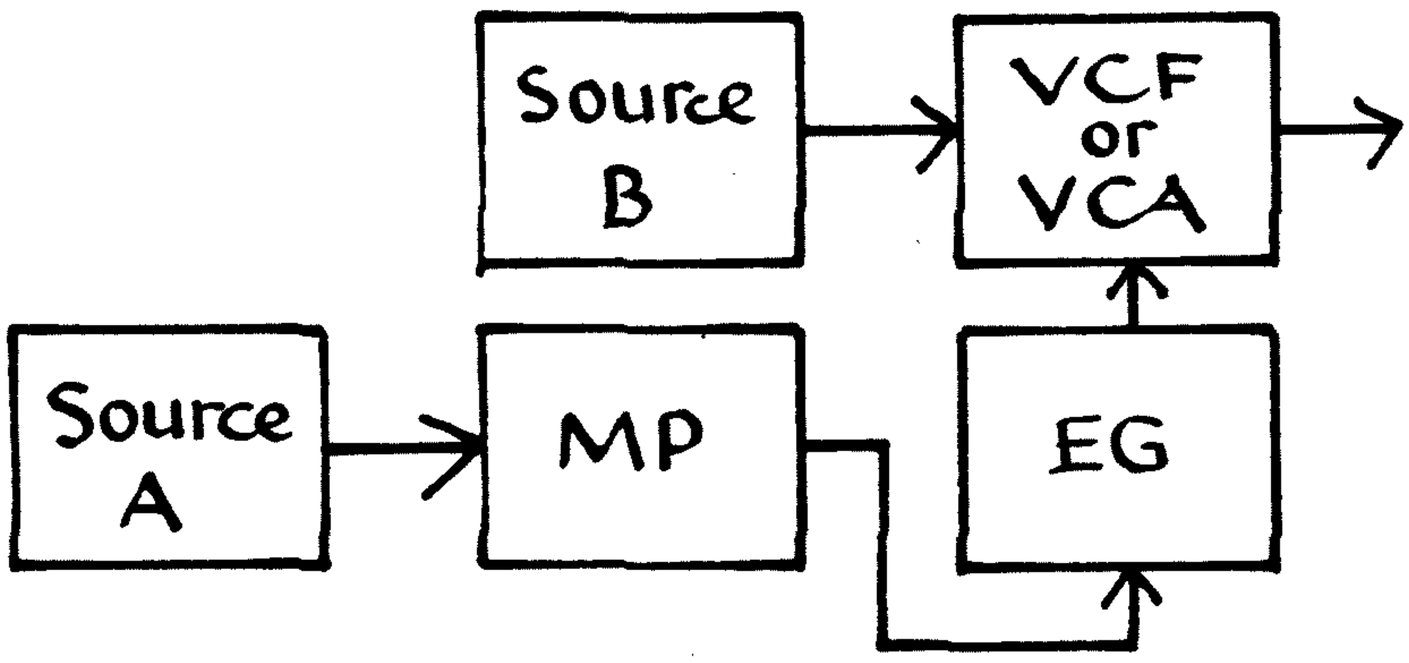
8.

We told you about this one in our comments on No. 7.

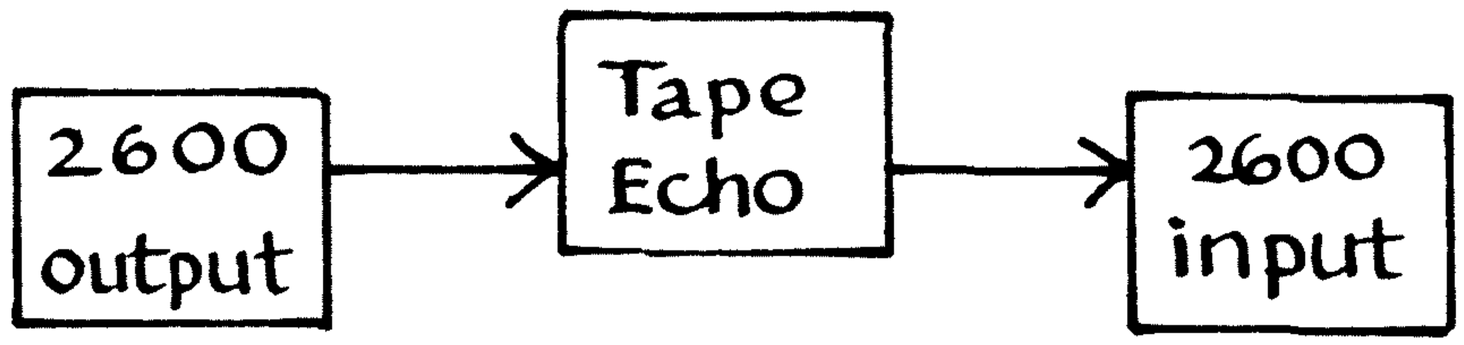


9.

We told you about this one, too, in our comments on No. 7. Possibly the most useful envelope would be the one from the AR generator, with maximum decay and minimum attack time. But try anything that occurs to you.

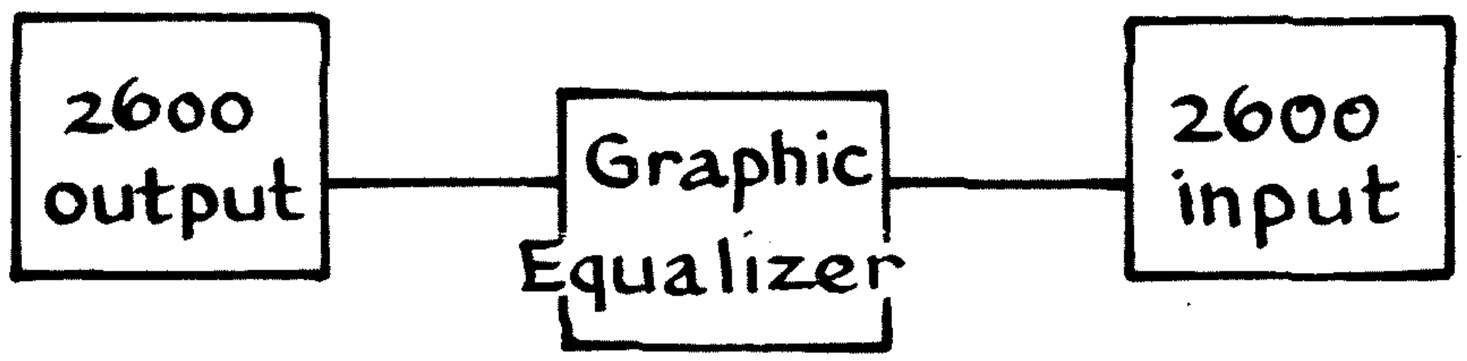


5.52 Externally processing internally-generated signals.



If you are deep enough into synthesis techniques to be familiar with tape head echo, you do not need any help from us.

COMMENT: same as No. 1.



6.0 ADJUSTMENT AND MAINTENANCE. Your Model 2600 was carefully adjusted and tested before it left the factory. Like any musical instrument, however, it will need periodic adjustment, tuning, and cleaning. If you follow carefully the procedures we give you in this section, you can bring your 2600 back to factory-perfect alignment and calibration whenever it becomes noticeably out of tune. In general, ARP Synthesizers are extremely stable and will not need tuning for weeks at a time or even longer, performance that is unequalled by any other synthesizer.

Maintenance is absolutely minimal. The only critical task here is occasionally to clean the keyboard contacts if they become obstructed by dirt or dust. Noncritical maintenance includes ordinary cleaning of control surfaces and protection from extremes of temperature and mechanical shock.

Should you experience any difficulty in operating the Model 2600, contact your dealer or the nearest factory-authorized distributor. He will be able to advise you. Refer to the TONUS warranty statement for information on servicing.

6.1 THE CALIBRATION AND ALIGNMENT PROCEDURES outlined in the following pages can be carried out without going inside the 2600 and without using any external test instruments other than your own ears. Adjustments are made by means of the semi-fixed trimmers recessed behind the small holes in the front panel. Use the small screwdriver supplied with your instrument.

If you are uncertain of your ability to perform any of these alignments without help, your nearest factory authorized servicing distributor will assist you or perform them for you without charge within the warranty period of 90 days; after that you will be charged for his assistance and/or service.

Note that throughout the alignment procedure the keyboard PORTAMENTO, TUNING, and TONE INTERVAL switches should always be in their right-hand position. The PANPOT slider may remain centered.

6.11 BEGIN with the VCO's. Each oscillator must be set so that its output frequency corresponds at least roughly with the markings on the INITIAL FREQUENCY slider and so that it tracks in unison with the other VCO's over a wide range of frequencies. In addition, the square-wave output from VCO-1 must be adjusted so that it is exactly symmetrical.

6.111 VCO FREQUENCY CALIBRATION.

1. Set all SLIDE CONTROLS to their MINIMUM, i.e. down or left position.
2. On the KEYBOARD, all switches should be set to their RIGHT-HAND position.
3. On VCO-1, the FINE TUNE slider should be centered, and the range switch set to AUDIO.
4. Insert a dummy plug or the end of a patch cord into the KBD CV jack input to VCO-1.
5. Now set the INITIAL OSCILLATOR FREQUENCY slider to the 100Hz panel marking.
6. Advance VCO-1 input on Filter to maximum; advance Filter INITIAL FILTER FREQUENCY to maximum; advance Filter input to Mixer, and Speaker Volume controls, so that a tone is audible.

7. ADJUST THE VCO-1 TRIMMER LABELLED "FREQ CAL" UNTIL PITCH IS SLIGHTLY FLAT OF A-110Hz, TWO OCTAVES BELOW A-440Hz. (A-440 is the first A above middle C.)
8. Repeat steps 3 through 7, substituting VCO-2 for VCO-1.
9. Repeat steps 3 through 7, substituting VCO-3 for VCO-1.

6.112 VCO TRACKING ADJUSTMENTS.

1. All sliders should be at their MINIMUM (down or left) position.
2. Insert a dummy plug into the KBD CV jack input to VCO-3.
3. Advance the VCO-3 input to the Filter, the Filter input to the Mixer, and the two Speaker Volume controls to a comfortable level.
4. Set the VCO-3 INITIAL FREQUENCY to 100Hz and the FINE TUNE to center.
5. Open partially the VCO-2 input to the Filter and, while holding down the lowest note on the keyboard, tune VCO-2 to unison with VCO-3 by "zero-beating" the two oscillators. (In using the FINE TUNE slider for precise tuning of this sort, the best method is to press in one direction with your thumb at the base of the slider, resting on the panel, while the index finger holds the top of the slider back in the other direction, as in the drawing.)
6. Holding down the C three octaves above the lowest note on the keyboard, ADJUST THE TRIMMER LABELLED "1V/OCT" UNTIL VCO-2 EXACTLY ZERO-BEATS WITH VCO-3's EIGHTH (which is three octaves above the fundamental) HARMONIC. This will be made easier if you adjust the Filter inputs so that VCO-3 is considerably louder than VCO-2.
7. Repeat steps 5 and 6 substituting VCO-1 for VCO-2.
8. Remove the dummy plug from the KBD CV input to VCO-3.
9. While holding down the lowest note on the keyboard, tune FCO's 2 and 3 to unison by zero-beating them.

10. Holding down the C three octaves above the lowest note on the keyboard, ADJUST THE VCO-3 TRIMMER LABELLED "1V/OCT" UNTIL VCO-3 ZERO-BEATS WITH VCO-2.
11. Insert a dummy plug into the KBD CV input to VCO-3, and set the VCO-3 INITIAL FREQUENCY control at 1KHz.
12. Holding down the lowest note on the keyboard, tune VCO-2 to unison with VCO-3 by zero-beating the two oscillators.
13. Holding down the C two octaves above the lowest note on the keyboard, ADJUST THE VCO-2 TRIMMER LABELLED "HI-FREQ TRACK" UNTIL VCO-2 EXACTLY ZERO-BEATS WITH VCO-3's FOURTH (two octaves above the fundamental) HARMONIC.
14. Repeat steps 12 and 13, substituting VCO-1 for VCO-2.
15. Remove the dummy plug from VCO-3's KBD CV input.
16. Holding down the lowest note on the keyboard, tune VCO-3 and VCO-2 to unison by zero-beating them.
17. Holding down the C two octaves above the lowest note on the keyboard, ADJUST THE VCO-3 TRIMMER LABELLED "HI-FREQ TRACK" UNTIL VCO-3 EXACTLY ZERO-BEATS WITH VCO-2.

6.113 VCO-1 SQUARE WAVE SYMMETRY ADJUSTMENT.

1. Begin with all sliders at minimum (down or left).
2. With a patch cord, connect the VCO-1 SAWTOOTH output to the VCO-1 input jack on the Filter. Raise the attenuator for this input to maximum.
3. Raise the VCF Mixer input to maximum and the Speaker Volume controls to maximum.
4. Set the VCO-1 INITIAL OSCILLATOR FREQUENCY control to a little under 1 KHz.
5. Set the INITIAL FILTER FREQUENCY control to 1KHz and the Filter RESONANCE control to just below oscillation.
6. Tune the Filter to the 2nd harmonic of the VCO-1 sawtooth. (If you have any difficulty, tune first to the fundamental and then find its octave. The Filter will peak sharply at each harmonic.)
7. REMOVE THE PATCHCORD CONNECTION AND ADJUST THE "SYMMETRY" TRIMMER UNTIL THE 2ND HARMONIC DISAPPEARS COMPLETELY.

6.12 ADJUSTING THE VOLTAGE-CONTROLLED FILTER. Here the "offset" adjustment minimizes any DC in the Filter output, and the "gain" adjustment sets the Filter gain to exactly unity. The frequency calibration and tracking adjustments are the same as for the VCO's but are somewhat less critical.

6.121 OUTPUT OFFSET ADJUSTMENT.

1. Begin with all controls at minimum (down or left).
2. With a patch cord, connect the VCO-2 SINE output to the Mixer VCA input. Raise the attenuator over this input to maximum and raise the Speaker Volume controls until the sine tone is comfortably audible.
3. With another patch cord, connect the VCF output to the first attenuator-governed FM input to VCO-2, interrupting the S/H connection to that input. Raise the attenuator over that input to maximum. You should hear a change in the pitch of the sine tone.
4. ADJUST THE "OUTPUT OFFSET" TRIMMER FOR MINIMUM PITCH CHANGE WHEN THE PATCH CORD CONNECTING THE FILTER TO VCO-2 IS ALTERNATELY REMOVED AND INSERTED AGAIN.
5. Remove all patch cords.

6.122 GAIN ADJUSTMENT.

1. Set all controls to minimum.
2. Make the following patch connections with patch cords: VCO-3 SAWTOOTH to VOLTAGE PROCESSOR INPUT No. 1, FIRST PROCESSOR OUTPUT to second MIXER INPUT (interrupting the VCA connection to that input).
3. Raise to maximum the VCO-3 input to the Filter, the Filter INITIAL FREQUENCY control, and the two Mixer input attenuators. Raise the Speaker Volume controls to a high listening level.
4. ADJUST THE "GAIN" TRIMMER FOR MINIMUM VOLUME THROUGH THE SPEAKERS. If necessary raise the Speaker Volume controls to maximum.
5. Remove all patch cords and return all controls to minimum.

6.123 FREQUENCY CALIBRATION.

1. Begin with all control sliders at minimum, i.e. down or left.
2. With a patch cord, connect the VCO-2 SINE output to the Mixer, interrupting the VCA connection.

3. Insert dummy plugs into the KBD CV input jacks on VCO-2 and on the Filter.
4. Set VCO-2 frequency at 1KHz; set the Filter frequency at 1KHz.
5. Advance the Filter RESONANCE to maximum.
6. Raise both of the Mixer input attenuators halfway. Raise the Speaker Volume controls to a comfortable level.
7. ADJUST THE "FREQ CALIBRATE" TRIMMER ON THE VCF UNTIL THE VCF AND VCO-2 OUTPUTS ZERO-BEAT.

6.124 1V/OCTAVE CALIBRATION.

1. Begin with all control sliders at minimum, i.e. down or left.
2. With a patch cord, connect the VCO-2 SINE output to the second MIXER input, interrupting the VCA connection to that input.
3. Set VCO-2 and the Filter both to about 500Hz.
4. Advance the Filter RESONANCE to maximum.
5. Open the Mixer input attenuators about halfway and set the Speaker Volume controls to a comfortable level.
6. Holding down the lowest note on the keyboard, tune the Filter to unison with VCO-2 by zero-beating them.
7. Holding down the C three octaves above the lowest note on the keyboard, ZERO-BEAT THE FILTER AND VCO-2 BY ADJUSTING THE "1V/OCT" TRIMMER ON THE FILTER.
8. Repeat steps 6 and 7 as needed for precise tracking.

6.13 ADJUSTING THE VOLTAGE-CONTROLLED AMPLIFIER.

Here we are concerned with high-frequency and control-signal rejection and with adjusting both the LINEAR and the EXPONENTIAL gain factor to unity.

6.131 HIGH FREQUENCY REJECTION.

1. Set all control sliders to their minimum, i.e. down or left, position.
2. With patch cords, connect the VCA output to the MIC PREAMP input, and connect the MIC PREAMP output to the PANPOT input.
3. Set the MIC PREAMP "RANGE" at x1000 and GAIN at maximum.

4. Set the VCO-3 frequency to 1KHz.
5. Raise the Filter input from VCO-3 to maximum, the Filter initial frequency to maximum, and the Filter input to the VCA to maximum. Center the panpot and raise the two Speaker Volume controls to maximum.
6. NOW ADJUST THE VCA TRIMMER LABELLED "HI-FREQ REJ" UNTIL THE HIGH FREQUENCIES HEARD THROUGH THE SPEAKERS ARE AT A MINIMUM.

6.132 CONTROL REJECTION.

1. Set all controls to minimum.
2. With a patch cord, connect the VCO-2 SINE OUTPUT to the VCA LINEAR CONTROL input. Set VCO-2 to 1KHz.
3. Raise the LINEAR CONTROL input attenuator to maximum and the Mixer VCA input to maximum, and the Speaker Volume controls to maximum.
4. ADJUST THE VCA TRIMMER LABELLED "CONTROL REJ" FOR MINIMUM SIGNAL THROUGH THE SPEAKERS.

6.133 GAIN ADJUSTMENTS.

1. Set all control sliders at minimum, i.e. down or left, position.
2. Advance Filter RESONANCE to maximum and INITIAL FREQUENCY to 1KHz.
3. With patch cords, connect the Filter output to INVERTER INPUT No. 1 in the Voltage Processor section; connect the inverter OUTPUT to the first MIXER input, interrupting the VCF connection to that input.
4. Raise to maximum both Mixer inputs and the Filter input to the VCA.

6.1331 LINEAR GAIN ADJUSTMENT: continued through from 6.133.

5. Raise the VCA INITIAL GAIN to maximum. Raise the Speaker Volume controls to a comfortable listening level.
6. NOW ADJUST THE VCA "LINEAR GAIN" TRIMMER FOR MINIMUM AUDIBLE SIGNAL THROUGH THE SPEAKERS. Raise the speaker volume controls to maximum if necessary.
7. Return the VCA INITIAL GAIN control to minimum.

6.1332 EXPONENTIAL GAIN ADJUSTMENT: continued through from 6.133.

5. Raise the EXPONENTIAL CONTROL input to the VCA to maximum.
6. On the ADSR Envelope Transient Generator, raise the SUSTAIN VOLTAGE control to maximum.
7. Holding down the lowest note on the keyboard, ADJUST THE VCA TRIMMER LABELLED "EXP'L GAIN" FOR MINIMUM SIGNAL THROUGH THE SPEAKERS.

6.14 ADJUSTING THE RING MODULATOR, for positive and negative nulling and proper gain.

6.141 NULL ADJUSTMENTS.

1. Set all controls to minimum.
2. Set VCO-1 to 1KHz.
3. Advance to maximum the VCO-1 input to the Ring Modulator, the RING MOD input to the VCA, and the VCA input to the Mixer. Raise the Speaker Volume controls to maximum.
4. ADJUST THE RING MODULATOR TRIMMERS LABELLED "POS NULL" AND "NEG NULL" ALTERNATELY, TUNING FOR MINIMUM SIGNAL THROUGH THE SPEAKERS.

6.142 GAIN ADJUSTMENT.

1. Set all controls to minimum.
2. With patch cords, connect the KEYBOARD CONTROL VOLTAGE output to the Ring Modulator left input, interrupting the VCO-1 connection to that input; connect the VCO-1 SQUARE output to the Ring Modulator right input, interruptingthe VCO-2 connection to that input; and connect the RING MODULATOR OUTPUT to VCO-2's unattenuated control input, interrupting the KBD CV connection to that input.
3. Set VCO-2 INITIAL FREQUENCY to about 500Hz.
4. Set the VCO-1 INITIAL OSCILLATOR FREQUENCY control to about 1KHz and switch it to its LOW-FREQUENCY range.

5. Set the Ring Modulator coupling to DC.
6. Advance to maximum: both Ring Modulator inputs, the Filter input from VCO-2, the Filter frequency, and the Mixer input from the Filter. Raise the Speaker Volume controls to a comfortable level.
7. Holding down the lowest F-sharp on the keyboard, ADJUST THE RING MODULATOR "GAIN" TRIMMER UNTIL THE TONE FROM VCO-2 IS SHIFTING EXACTLY ONE OCTAVE.

6.15 This concludes the calibration and alignment procedure.

6.2 MAINTENANCE. Most of what you should be concerned with here is merely preventive: do not expose your Model 2600 to extreme heat or cold, don't leave it out in the rain, or set it down in the middle of busy intersections.

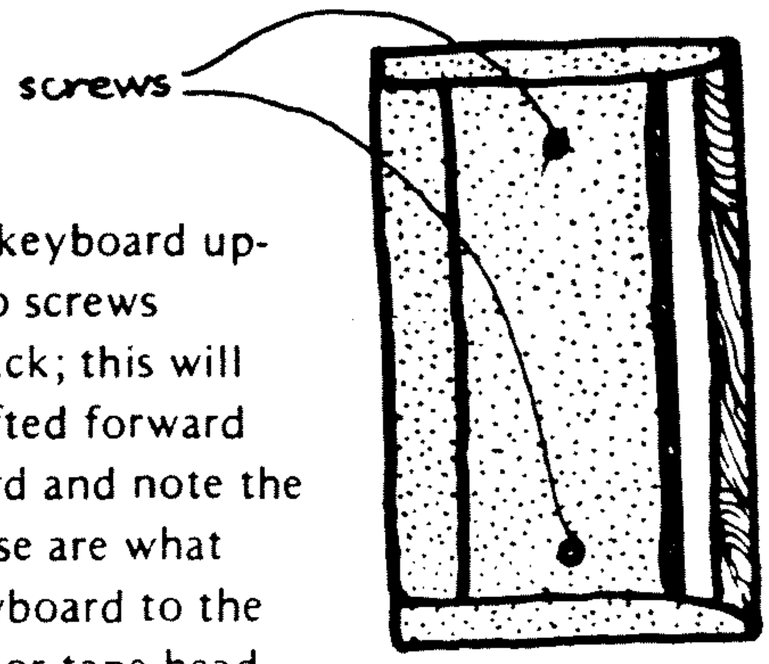
If after extended use the exposed surfaces of the 2600 begin to get dirty, they may be cleaned with a soft damp cloth. If you decide to wax it, take great care to prevent the wax from entering the control slider slots where it could build up and eventually affect the sliders' contact surfaces. Take care in general that wax does not build up in any crevices, or between the keys of the 3604 controller.

Don't try to take the 2600 apart to get inside; doing so will void your warranty. All possible interconnections can be made externally; we have designed the 2600 specifically so that all user-adjustable trimmers are available without disassembly.

DO NOT ATTEMPT TO ADJUST THE FIXED "PRESET" TONE INTERVAL TRIMMER ON THE 3604 KEYBOARD. Unlike the other exposed trimmers, it cannot be precisely trimmed without external test equipment and is factory-preset for EXACTLY a +4V interval across the keyboard span. This interval in fact serves as a reference for much of the alignment procedures in 6.1, and if it is disturbed the alignment procedures are worthless.

6.21 CLEANING THE KEYBOARD CONTACTS.

Occasionally the keyboard contacts will become obstructed by dust or dirt, causing the keyboard control voltage to become unsteady or intermittent. They may be cleaned with q-tips and alcohol or tape head cleaner.



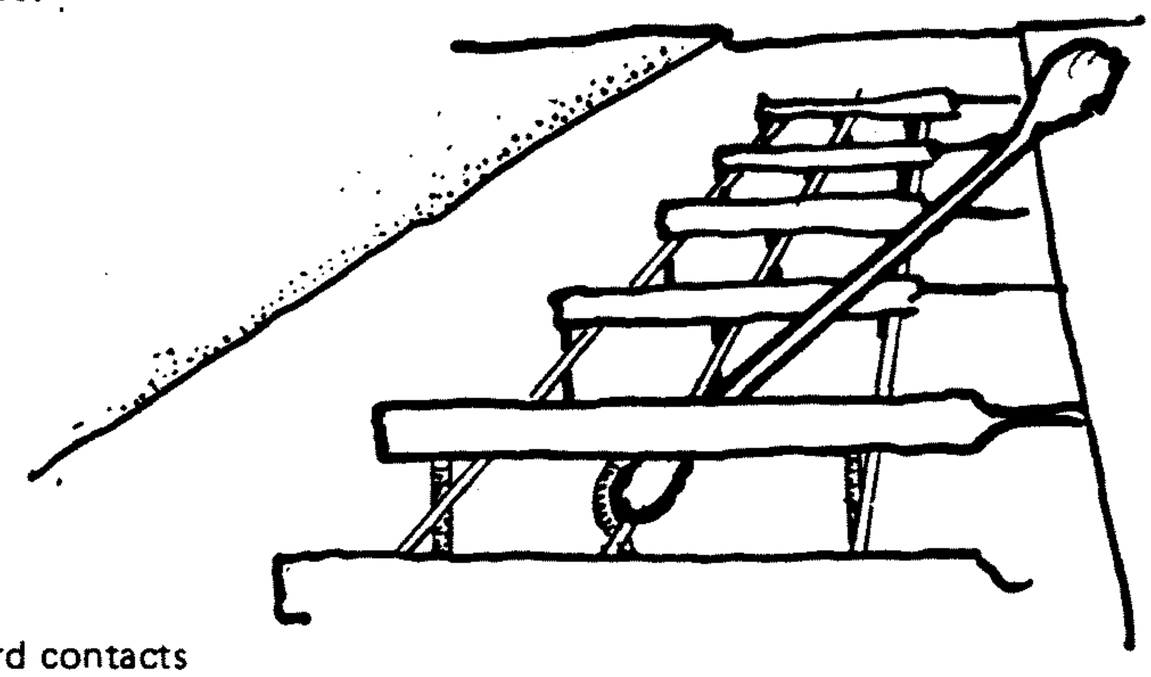
Underside of keyboard

To get access to the keyboard contacts, turn the keyboard upside down on a cloth-covered surface and remove the two screws indicated in the diagram. Push the cover panel slightly back; this will free its forward edge, and the whole panel can then be lifted forward and up. Press up one of the keys underneath the keyboard and note the three contacts that move when the key is depressed. These are what must be cleaned. Work carefully from one end of the keyboard to the other, inserting a q-tip moistened with isopropyl alcohol or tape head cleaner between each spring contact and its bus wire. Then depress the key associated with that contact several times in succession. Access to the black keys may be gained by supporting only the two ends of the keyboard, on books or something, so that the keyboard is raised above your working surface. After doing several contacts in this way, go back and with a dry q-tip scrub dry the residue from the contacts you have cleaned. Then go on to another group of three or so keys.

Take care not to bend the bus wires that run the length of the keyboard; they are adjusted so that all three contacts, when a key is depressed, are made as nearly as possible simultaneously. For the same reason, take care that the contact springs do not become stretched and distorted.

6.22 ELECTRICAL PRECAUTIONS IN CONNECTING EXTERNAL EQUIPMENT: in the normal course of events it is not likely that any special precautions need be taken. The high-impedance inputs to the 2600 functions will accept up to 50V RMS signals without overload, though a more reasonable limit—say 40V RMS— should be imposed for complete assurance of safety. Both inputs and output may be paralleled or shorted without damage, though a single VCO output, for example, if connected to enough separate inputs, may be loaded down enough to produce a perceptible pitch change, this does not in any way damage the VCO.

Because of the high input impedance of any 2600 functional circuit, it may, for example, be bridged directly across the speaker coils of electric guitar amplifiers etc.


Cleaning the keyboard contacts

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